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M2FEC: An effective FEC based multi-path transmission scheme for interactive multimedia communication

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ABSTRACT

The oncoming prosperity of interactive multimedia application triggers significant challenges to current best-effort Internet due to such application's stringent delay, loss and bandwidth requirements, and Internet's unpredictable dynamics. Multi-path transmission and error-resilient coding are two promising approaches to alleviate these problems. This paper attempts to introduce error-resilient coding into multi-path transmission to better trade off between multi-path bandwidth resource consumption and reliable media quality. We propose a model for multi-paths interactive multimedia transmission and develop M2FEC—a FEC based transmission scheme which maximizes the overall quality at the client under various constraints based on the proposed model. Numerical simulation and PlanetLab experiments demonstrate the effectiveness and practicability of M2FEC in theory and in empiricism, respectively.

1. Introduction

With the rapid development of multimedia and network technologies, various interactive multimedia applications are rising up over current Internet, including voice/video over IP, video conference, massive multi-player online games, distance learning and tel-medicine. These commercioganic applications allow people to interact regardless of geographical distance between them. They are changing the way people communicate and access information. The high interactivity and bandwidth consumption properties of such applications bring forward stringent delay, bandwidth and packet loss requirements. However, current generation Internet was not originally designed for real-time applications but for data transmission and only provides best-effort services, which makes the deployment of high-quality interactive applications unprecedented challenge.

During the past decade, several efforts have been devoted to provide QoS for Internet and improve end user experience. The existing approaches can be classified into three categories. From IP-layer perspective, InterServ, DiffServ, RSVP and MPLS [1–4] have been proposed. However, the need of changing network infrastructures or service model limits the wide deployment for these solutions. From application-layer perspective, overlay routing techniques have been widely used in practice to overcome Internet's inherent limitations and enhance end-to-end performance for interactive applications [5–8]. Besides the advantage of the potential performance gain, robustness improvement and flexibility routing, overlay routing also has the great benefit of simple to implement compared with IP-layer solutions. From coding perspective, error-resilient techniques such as layered coding (LC) [9], multiple description coding (MDC) [10,11] and forward error correction (FEC) [12] have been proposed to deal with the heterogeneity and time varying nature of the Internet.

No need for special support and easy deployment make application-layer solution and coding layer solution very popular in current Internet. However, most of the previous research focuses on high-quality overlay path selection or efficient coding algorithms design, few paying attention to effectively utilize selected overlay paths with existing error-resilient coding for interactive applications in particular. This paper aims to combine two promising technologies, multiple overlay paths transmission in application-layer solution and FEC coding in coding layer solution, to better trade off between bandwidth efficiency and end-to-end path reliability, thus providing more pleasant experience for end users.

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Specifically, the main contributions in this paper include twofold. First, develop an explicit rate–distortion (R–D) based analytical framework M2FEC that models FEC based multi-path transmission for interactive Multimedia. Optimal its coding encoding bitrates, FEC redundancy and multi-path scheduling strategies under a given network environment by addressing the proposed optimization. Secondly, evaluate the efficiency of M2FEC theoretically and empirically by comparing with several other transmission schemes such as Mesh, Round-Robin, Single path FEC and Single path. Implement a practical online computation module of M2FEC transmission scheme on PlanetLab testbed.

The remainder of the paper is organized as follows. Section 2 outlines current multi-path research and describes the motivation of M2FEC. Section 3 proposes M2FEC transmission model in detail. Numerical analysis results are presented in Section 4. PlanetLab evaluation results are described and illustrated in Section 5. Finally, Section 6 concludes the paper with a discussion of future work.

2. Motivations

2.1. Analysis of existing multi-path overlay routing

Traditionally, sender transmitted packets to receiver through a single path selected by IP routing protocol. A variety of studies have revealed that many of the current Internet's routes are far from reliable [13,14]. If network is congested along the default IP-layer path, multimedia delivering will suffer from high loss rate and jitter which will decrease its consistency and interactivity. The emergence of overlay techniques [15] triggers research on sending packets simultaneously over multiple paths as a diversification scheme to combat the unpredictability and congestion in the Internet. To create a redundant path in the overlay framework, the sender sends packets to a relay node, and the relay node then forwards packets to the receiver. If one path between a particular sender and receiver experiences packet loss due to congestion, packets traversed through the other path can be used to recover the lost packets.

Many multi-path routing algorithms have been proposed to find appropriate relay nodes. Some approaches suggest increasing path diversity under the heuristic of topology, such as earliest divergence rule (EDR) [16], disjoint multi-path QoS routing (DMQR) [17] and widest disjoint paths (WDP) [18]. Their basic idea is to establish overlay path which takes a different underlying physical routine from default IP-layer path between the sender and receiver. Other work focuses on improving the metrics of selected overlay paths to better satisfy the QoS requirements of applications, including reliability [14], end-to-end delay [6], available bandwidth [8,27] and compound metrics [5,28].

Empirical study in literature [7] shows the benefit of two special multi-path deliver schemes: Mesh (which duplicates all the packets along different paths) and Round-Robin (where each path carries equal traffic and packets are transmitted in a Round-Robin manner). Mesh guarantees reliable quality improvement at the cost of doubled bandwidth consumption, while Round-Robin makes full use of the bandwidth resource at the probability of suffering from heavy packet loss. In addition, these two schemes do not consider behavior differences between paths and treat every path equally. Literature [12,29] introduces FEC into multi-path transmission, however this work focuses on packet loss instead of media quality, and FEC related parameters are fixed in all scenarios without considering media encoding rate and dynamics of network environment.

Inspired by above work, this paper attempts to propose an analytical model that can estimate the expected media quality quantitatively when overlay paths have been selected by certain overlay routing algorithms and transmission scheme is given. By optimizing the proposed model, appropriate transmission parameters such as FEC redundancy, encoding bitrates and scheduling strategies which maximize the prospective media quality under current network environments and application requirements could be given.

2.2. M2FEC design requirements

Before expatiating on M2FEC, we first state the following design requirements:

(1) *Optimization of transmission scheme:*
This requirement is the fundamental objective of this paper, which has already been described in the above section.

(2) *Consideration of the nature of interactive applications:*
From the transport layer point of view, interactive communication differs significantly from traditional data applications. First, traditional applications use TCP as their transport layer protocol in which transmission rate of sender is determined by the level of congestion. Interactive applications, on the other hand, usually react to network congestion with less flexibility, due to their relatively constant transmission rate requirement. Secondly, interactive applications have stringent timing constraints. ITU-T Recommendation G.114 specifies that one-way transmission delay of VoIP should be better below 150 ms, and delay above 400 ms is unacceptable [19]. As video should be synchronous with voice during communication, video based interactive application has similar delay requirement with VoIP. In addition, in order to faithfully recreate the original data and play back packets continuously, packets must arrive at the receiver before a fixed delay offset, otherwise late arrival packets will be considered "lost" at the receiver side. Thirdly, interactive applications can tolerate a small amount of packet loss without significant degradation of user experience, so that reliable data transfer is not absolutely critical.

(3) *Resilience to Internet dynamics:*
Current Internet end hosts interconnect with each other through a myriad of switches, routers and different protocols. These components have a wide range of speeds, buffering capabilities and routing policies. The need for communication across different administrative domains and the evolution of technologies have required the need of inter-operability, which aggravates the heterogeneity of the Internet. Therefore behaviors of individual end-to-end path vary violently and unpredictably in time domain. A good transmission scheme should have the ability to bear delay, packet loss and available bandwidth fluctuation.

(4) *Practicability of implementation:*
To ensure the practicability of M2FEC, the following principles should be followed. First, input network parameters in the model can be measured efficiently with current Internet measure technology and tools. Secondly, the model should not be too complex and could be calculated rapidly for real-time make-up and update of transmission scheme. Thirdly, to avoid concussion, slight network parameters variety should not change the model's output results significantly.

3. M2FEC transmission model

3.1. Assumption and notations

We begin the description of M2FEC with a set of assumptions:

(1) The original real-time streams of the same pair of nodes is generated and encoded at uniform speed across time.

(2) The source-encoded streaming is packetized into the same size packets, and each packet is sent in a time slot of fixed duration.
(3) By certain multi-path overlay routing algorithm such as [17,28], sender and receiver have already select several overlay paths with good performance between them, and this paper only investigates the coding and transmission strategies between the communication pairs.
(4) The loss and delay varying behaviors of the two selected paths are approximate uncorrelated with each other. Note that this hypothesis is not always true but highly depend on the degree of their physical link disjointness.

Table 1 summarizes the basic notation used in this paper for easy reference. The subscript i represents the parameters of the ith path, and path parameters have no subscript when using single path transmission only or the parameters among all the paths are exactly same.

3.2. Rate–distortion (R–D) model

R–D theory was created by Shannon in his foundational work on information theory. In R–D theory, rate R denotes the number of bits per data sample to be stored or transmitted; distortion D is defined as the variance of the difference between input and output signal (i.e., MSE—mean squared error). R–D theory provides the theoretical foundations for lossy data compression which addresses the problem of determining the minimal amount of entropy R that should be communicated over a channel, so that the input signal can be approximately reconstructed at the output signal without exceeding a given distortion D.

Let $\{X_k\}_{k=1}^{M}$ denotes a sequence of source symbols input to source encoder, and $\{\widehat{X}_k\}_{k=1}^{M}$ denotes the reconstructed signal after source decoding. Then the average distortion (MSE) will be:

$$D = E\left\{\frac{1}{M}\sum_{k=1}^{M} d(X_k, \widehat{X}_k)^2\right\} \tag{1}$$

where $d(,)$ is the Euclidean distance between X_k and $\widehat{X}_k$.

To estimate the reconstructed media quality, namely average distortion, this paper incorporates analytic model recently developed in [30], which can fit scalable video codec such as MPEG-4 FGS quite well. The expression of this cited model is:

Table 1
Notations and meanings.

Notation	Meaning	Default value and unit
p_i	Packet loss rate along the ith path without FEC coding	N/A
p_F	Packet loss rate with FEC coding. It is a function of p_i and FEC protection levels	N/A
l	Bytes of payload in a packet	byte
h	Bytes of header per packet	54 bytes
$L_{\max}$	Maximum allowed packet size	560 bytes
$RS(n,k)$	Reed–Solomon (RS) code parameter, (k data packets and $n-k$ redundant packets in a block)	N/A
B_{a_i}	Maximum bandwidth allocated along the ith path	kbps
B_i	Actual source-encoded bitrates assigned on the ith path	kbps
t_i	One-way end-to-end delay of the ith path	100 ms
t_r	One-way interactive application's delay requirement	300 ms
R	The source-encoded rate in terms of the number of bits per source pixel	b/p
N	Number of the paths	1 or 2
D	Average distortion	N/A
a,b,c	Parameter in R–D model	N/A
α	Penalty factor for irrecoverable packets	N/A
$W \times H$	Resolution of video frame (in pixels)	352×288 p/f
F	Frame rate	25 f/s
C	A known constant that depends on the chroma subsampling format	1.5 (4:2:0 format video)

$$D(R) = 2^{aR+b\sqrt{R}+c} \tag{2}$$

where a, b, c are parameters. For offline-encoded video sequences, the value of these parameters can be pre-computed. For real-time encoding, initial values are assumed according to the characters of communication scenarios, and these values are refined as more frames are encoded.

Consider the illustrative R–D curves in Fig. 1. The actual R–D curves are produced by encoding CIF format test sequences Foreman and Coastguard by a standard MPEG-4 encoder FFmpeg [31] with default parameters. These empirical results demonstrate that the R–D model of formula (2) fits the actual R–D curves quite well. Compared with other R–D models (e.g., [20,21,32]), it makes a relatively better trade off between estimation accuracy and calculation complexity.

Considering a transmission with FEC coding, the expected distortion at the receiver will be the weighted average of packets successful recovered scenario and packets failure recovered scenario. If packets irrecoverable probability after FEC decoding is p_F, the weighted distortion will be:

$$D = (1-p_F)2^{aR+b\sqrt{R}+c} + \alpha p_F 2^{c} \tag{3}$$

where α is a penalty factor for packets that can not be recovered. The symbol p_F and R in formula (3) will be derived in the following sections. α can be calculated by assuming an average value for each pixel and then computing the expected error.

3.3. Effects of FEC on packets loss

FEC is an error correction coding method which allows the receiver to detect and correct errors without interactions by adding redundant information at the sender side. Compared with another widely used error correction method retransmission, FEC reduces the time for recovering missing packets at the expense of bandwidth consumption by redundant packets. So this technique is quite attractive for interactive multimedia. Reed–Solomon (RS) code is the most widely used FEC coding method in data storage such as CDs & DVDs and in data transmission such as DSL and WiMAX. A $RS(n,k)$ coding means k packets of source data are encoded at the sender to produce n packets of encoded data in such a way that any subset of k encoded packets suffices to reconstruct the source data. Such a code allows the receiver to recover from up to $n-k$ losses in a group of n encoded packets. It could be proved that irrecoverable probability with $RS(n,k)$ under Bernoulli packet loss will follow formula (4) (proved in Appendix A).

$$p_F = \frac{1}{k}E[Y] = p - \sum_{i=1}^{n-k}\binom{n-1}{n-k-i}(1-p)^{k+i-1}p^{n-k-i+1} \tag{4}$$

where p is the average packet loss rate along the path and the only parameter characters Bernoulli process. In Bernoulli process, packet losses are independent and identically distributed, however, it is well known that packet losses are correlated and often occur in bursts. Several models have been established to capture the stochastic characteristic of the underlying packet loss process of the Internet traffic, including Gilbert model [22], general k order Markov model [23] and Hidden Markov model [24]. This paper chooses Bernoulli model as there is only one parameter p in this model and average packet loss rate can be collected easily and quickly. Actually it will further improve the performance of M2FEC if more accurate packet loss model is employed into our framework.

3.4. Constraints

This section presents several constrains to make our model better satisfy interactive media's special requirements and tolerate Internet's dynamics described in Section 2.2.

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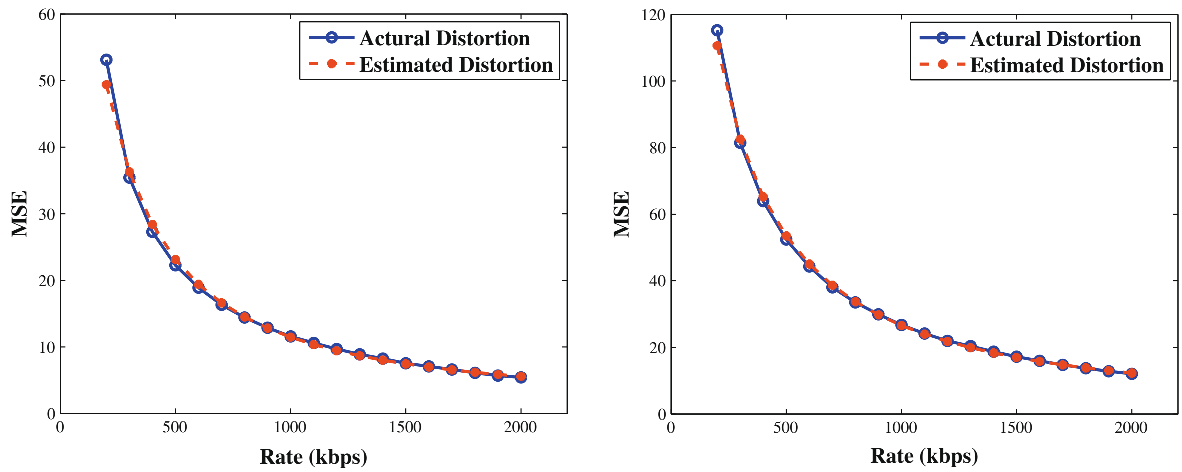


(a) Foreman (b) Coastguard

Fig. 1. Actual and estimated R–D curves.

First, interactive media applications have stringent end-to-end delay requirement. In FEC coding, the $n-k$ redundant packets in each block only can be computed after the k original packets have been generated, the delay of which could not be ignored in real-time application. Let t_f represents the time when the first packet in a FEC block is generated by the sender, and t_l represents the time when the last packet in the same FEC block is received by the receiver. To guarantee every packet in the same FEC block can be recovered on time, we require $t_l - t_f \leqslant t_r$, where t_r is the application's delay requirement. According to assumption (1) and (2) in Section 3.1, this inequality is equivalent to:

$$\frac{k}{B_i/l}\left(1+\frac{n-k}{n}\right)+t_i \leqslant t_r \tag{5}$$

where $B_i = R_i \times (W \times H \times F \times C)$, is the bandwidth consumption on ith path after source coding and before channel coding by FEC, l is the length of payload in every packet, and t_i is the one-way end-to-end delay of the ith path. As t_i varies across time, we sets one-way delay requirement t_r to be 300 ms instead of 400 ms declared in the ITU standard G.114 to adapt the dynamics of Internet delay.

Secondly, we desire FEC coding should not consume more bandwidth resource than other scheme such as Mesh and Round-Robin for fair comparison, and require the following inequality:

$$(h+l)(B_i/l)\frac{n}{k} \leqslant B_{a_i} \tag{6}$$

where h is the length of header in every packet, including RTP header, UDP header, IP header, etc., and B_{ai} is the maximum bandwidth allowed to consume along the ith path. To avoid overloading the path and make transmission resilient to background traffic variation, B_{a_i} is only limited to a small proportion of the total bottleneck bandwidth along the path (1/20 in our experiment).

Thirdly, considering MTU limitation and some routers assign low priority to large packets, we constrain packet size by inequality (7):

$$h+l \leqslant L_{\max} \tag{7}$$

3.5. M2FEC model

In the proposed M2FEC transmission, optimizational redundancy is added to the original source-encoded stream by FEC coding, then the FEC encoded and packetized stream is distributed among several overlay paths, finally packets reordering and decoding are processed at the receiver side. Base on the above analysis in this section, M2FEC can be concluded as the following optimization problem:

$$\min\left\{D=(1-p_F)2^{aR+b\sqrt{R}+c}+\alpha p_F 2^c\right\}$$
$$\text{s.t.}\begin{cases}\frac{k}{\sum B_i/l}\left(1+\frac{n-k}{n}\right)+\max(t_i)\leqslant t_r\\ (h+l)(B_i/l)\frac{n}{k}\leqslant B_{a_i}\\ B=R\times(W\times H\times F\times C)=\sum B_i\\ h+l\leqslant L_{\max} & i=1,\ldots,N\\ p=\frac{1}{\sum B_i}\sum B_i p_i\\ p_F=p-\sum_{j=1}^{n-k}\binom{n-1}{n-k-j}(1-p)^{k+j-1}p^{n-k-j+1}\end{cases} \tag{8}$$

where t_i, B_{ai} and p_i are input network parameters; FEC redundancy (n,k), aggregate source-encoded bitrates (B) and transmission parameters (B_i, l) are output transmission scheme; Expected media distortion (D) is the optimization objective. If $n=k=1$ and $B_1 = B_2 = \ldots = B_N$, Mesh and Round-Robin can be specialized as (9) and (10), respectively:

$$\min\left\{D=(1-p)2^{aR+b\sqrt{R}+c}+\alpha p 2^c\right\}$$
$$\text{s.t.}\begin{cases}\frac{1}{B/l}+\max(t_i)\leqslant t_r\\ (h+l)(B/l)\leqslant \min(B_{a_i})\\ B=R\times(W\times H\times F\times C)=B_i & i=1,\ldots N\\ h+l\leqslant L_{\max}\\ p=\prod p_i\end{cases} \tag{9}$$

$$\min\left\{D=(1-p)2^{aR+b\sqrt{R}+c}+\alpha p 2^c\right\}$$
$$\text{s.t.}\begin{cases}\frac{1}{B/l}+\max(t_i)\leqslant t_r\\ (h+l)(B_i/l)\leqslant \min(B_{a_i})\\ B=R\times(W\times H\times F\times C)=\sum B_i & i=1,\ldots,N\\ h+l\leqslant L_{\max}\\ p=\frac{1}{N}\sum p_i\end{cases} \tag{10}$$

Similarly, if the number of the paths N equals to 1, formula (8) can be specialized as single path transmission with or without FEC easily.

4. Numerical analysis

Section 3.3 models the transmission schemes of single path, single path with FEC, Mesh, Round-Robin and M2FEC as optimization problems. To evaluate the efficiency of M2FEC, this section will provide quantitative answers to the following three questions:

(1) Compared with the other four schemes, whether and how much M2FEC could improve media quality under various network environment?
(2) How much can M2FEC outperform the transmission schemes with other FEC redundancy?
(3) How well can M2FEC adapt to Internet unpredictable dynamics in theory?

To answer these questions, we set several typical network scenarios, and compare the expected media quality by solving the proposed model in Section 3. Optimization objective distortion (D) is converted to PSNR by formula $PSNR = 10\log_{10}\left(\frac{255^2}{MSE}\right)$ to make the comparison results intuitively and meaningfully. In particular, we only simulate two paths in Mesh, Round-Robin and M2FEC scheme because most of the multi-path routing algorithms only pick up two paths for low complexity and good path diversity consideration.

4.1. Distortion improvement overview

In this numerical analysis, packet loss ratio of each path ranges from 0 to 0.1 synchronously. The parameters in the R–D model formula (2) are acquired by fitting the two standard test sequence Foreman and Coastguard. Generally speaking, M2FEC performs best, followed by SPath-FEC, Mesh, Round-Robin and SPath (see Fig. 2). In pace with the increasing of packet loss, the two FEC schemes (SPath-FEC and M2FEC) explicitly slow down the media quality degradation speed than the two schemes without any redundancy (SPath and Round-Robin). Mesh behaviors approximatively similar to SPath-FEC, a little better than it when few packet losses occurs, while worse than it under heavy packet losses. However, Mesh achieves this error resilience at the cost of taking up more bandwidth resource with 100% redundancy, comparing with the redundancy of 50% in SPath-FEC in the worst case.

In fact, we conducted a large number of experiments and obtained similar results to the curves shown in Fig. 2. Due to space limitations, only parts of the results are summarized in Table 2, which shows M2FEC improves PSNR significantly by over 10 dB than SPath.

Table 2
Average PSNR (dB) in numerical simulation.

Sequence	B_{a_i} (kbps)	SPath	SPath-FEC	Mesh	Round-Robin	M2FEC
Foreman	200	21.5	29.7	28.8	22.0	32.0
Foreman	600	22.2	33.7	31.1	22.6	36.8
Foreman	1000	22.5	36.0	32.2	22.8	39.2
Coastguard	200	20.7	26.4	26.2	21.3	28.6
Coastguard	600	21.6	30.1	28.9	22.1	33.3
Coastguard	1000	22.0	32.4	30.3	22.4	35.8

4.2. M2FEC vs. regular FEC scheme

This section study the effect of FEC coding scheme with different parameters. Fig. 3 indicates the expected distortion (in dB) as a function of FEC coding parameter k and n. The points $n = k$ corresponding to the case when no FEC is used. Other simulation parameters are indicated in the figure caption. This figure illustrates that FEC coding can significantly improve the distortion performance compared to the case when no FEC is used given the appropriate coding parameters are chosen. For example, when no FEC is used,

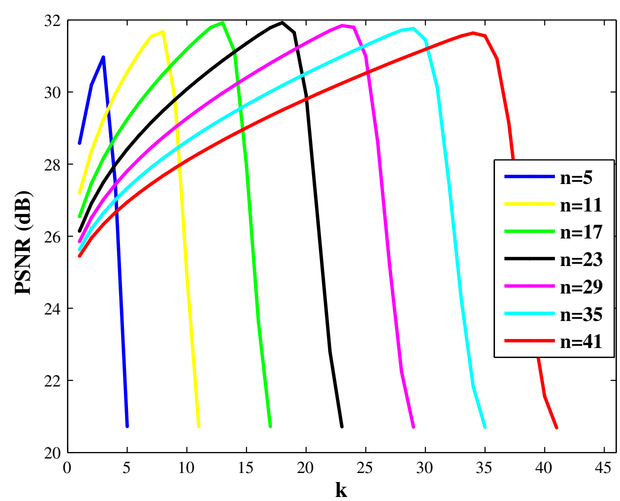
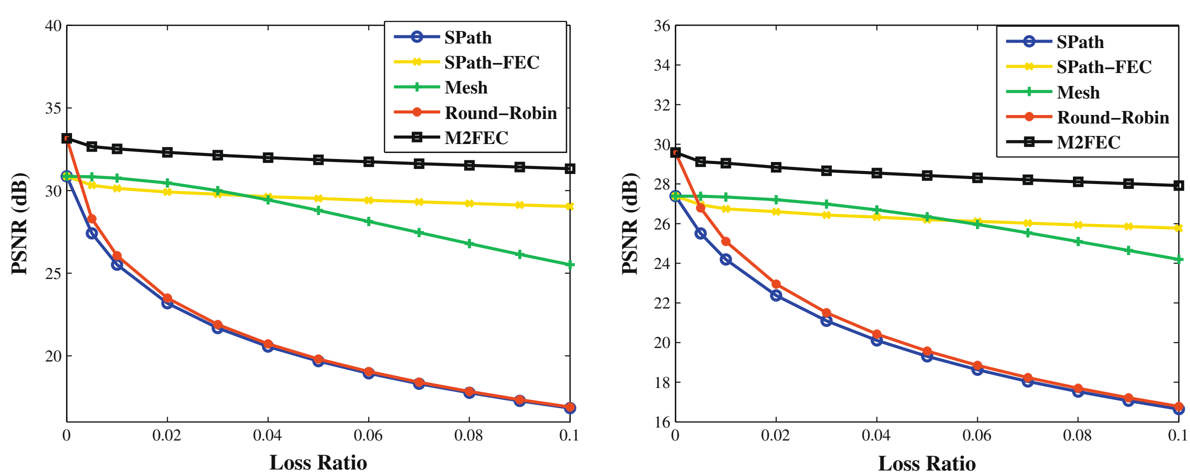


Fig. 3. Expected PSNR as a function of different FEC redundance ($B_{a_i} = 200$ kbps, $t_i = 100$ ms, $p_i = 0.04$).



(a) Foreman (b) Coastguard

Fig. 2. Numerical simulation curves with $B_{a_i} = 200$; kbps,$t_i = 100$ ms.

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the distortion is only about 20.5 dB, while FEC coding with $n = 23$ and $k = 19$ can improve it to 32 dB. Fig. 3 also indicates that for each coding block size n, there is an optimum k which can achieve the optimum performance.

Fig. 4 demonstrates a numerical example to illustrate the superiority of M2FEC to regular FEC scheme wih fixed parameters of $RS(23, 18)$. When packet loss ratio is lower than 3%, regular FEC scheme over-protects the source-encoded stream and waste some of the bandwidth resources; while when packet loss goes up, regular FEC is inadequate to combat the disgusting network condition, and PSNR will decrease rapidly by over 10 dB in the worst case. In contrast, M2FEC adaptively adjusts its parameters according to the diversification network condition guarantees performance in all the scenarios.

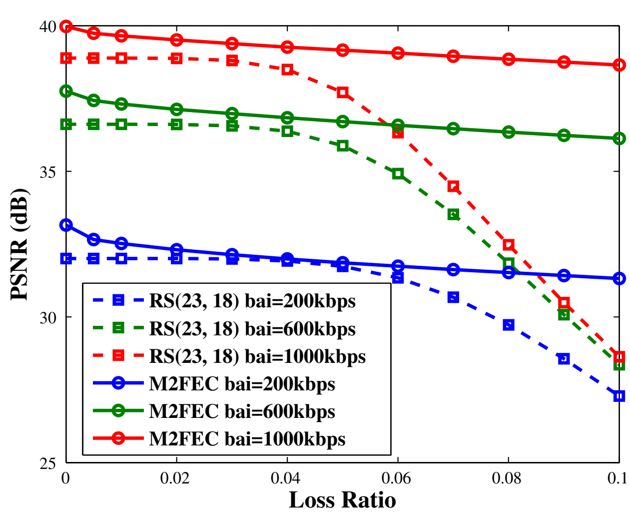

Fig. 4. Comparison of M2FEC and regular FEC scheme.

4.3. Dynamics adaptability

As stated in Section 2.2, current Internet's behavior varies significantly across time. In this section, we introduce some perturbation during the transmission simulation to investigate the dynamics adaptability of M2FEC.

According to the work in [25], 84% of end-to-end delay processes a shift Gamma-like shape with heavy tail. Gilbert model gives an acceptable level of accuracy to packet loss burst [23]. Gilbert model can be represented by a discrete-time two-state Markov chain, where current state of the stochastic process only depends on the previous value. There are two states—Good and Bad in Gilbert model. If the process is in Good state, the receiver receives all packets; If the process is in Bad state, all packets are lost. The process is characterized by the rates at which the Markov chain changes from "Good" to "Bad" state and vice versa. We use $Gama(4, 5)$ and Gilbert model $p_{good->bad} = 0.017, p_{bad->good} = 0.8$ in the simulation. As shown in Fig. 5, both M2FEC and Spath-FEC are resilient well to the given perturbation. The standard deviation decreases 70% by M2FEC and 60% by Spath-FEC than the two non-redundancy schemes, respectively, and even performs 30% and 15% better than the 100% redundancy scheme Mesh.

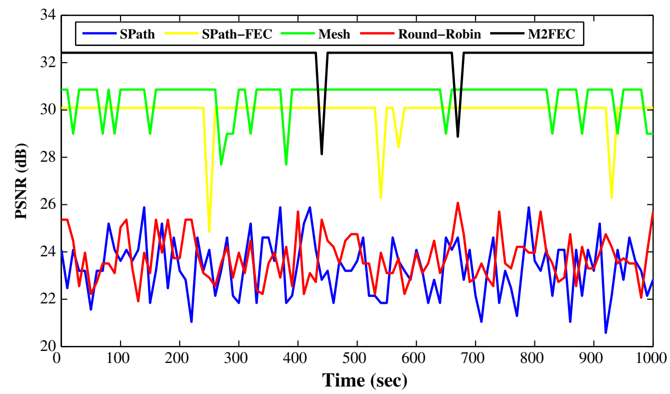

Fig. 5. PSNR fluctuation when end-to-end delay displays a Gama-like distribution $Gama(4, 5)$, and packet loss performs like Gilbert model. $B_{a_i} = 200$ kbps, $t_i = 100$ ms, $p_i = 0.02$.

5. Internet experiment

5.1. Experiment setup

To evaluate the effectiveness of M2FEC in actual Internet communication, we implemented an online transmission module on PlanetLab (see Fig. 6). There are three types of nodes in the module—sender, forwarder and receiver. The senders collect round-trip delay and packets loss rate of each path with the cooperation of forwarders and receivers. Maximum bandwidth along each path is allocated according to the information provided by the project of S3 (Scalable Sensing Service) [26], following the formula of $\min(Bw\_bottleneck/20, Bw\_available/5)$. After calculating transmission parameters by addressing proposed optimization in Section 3, standard CIF format source sequence is encoded by MPEG-4 codec FFmepg with groups of pictures (GOP) size of 15 frames. Next by FEC encoding and packetization, RTP packets with timestamp and sequence number are sent out directly to the receiver, or to the forwarder which relays the packets to the receiver, according to the senders' scheduling. At the receiver side, the original stream can be reconstructed by reordering and decoding received packets. Considering that interactive multimedia has stringent delay requirements, packets arrive beyond 400 ms one-way end-to-end delay is considered to exceed the receiver's playout time and will be dropped in our experiment.

We implement the above module to 20 PlanetLab nodes, which are scattered in 10 countries and 3 continents (see Appendix B). Through only a median scale testbed, our experiments contain serval topical long-distance communication, such as Asia to North American, Europe to North American, Asia to Europe, Education network to Commercial network, and so on. The forwarders are selected by the algorithm proposed by our group' previous work [28], which can provide excellent delay and path diversity at the same time. In fact, increasing the scale of our testbed tends to benefit the quality of overlay paths as the communication pairs will have more choices, thus helps to improve the media quality finally.

The series of experiments concentrate on video because it places the greatest load on the network in terms of bandwidth, as well as delay and packet loss. It is not difficult to extend the results of this work to voice. Two test sequences are used during the experiment. One is Foreman—a head-and-shoulders type sequence applies to simple video calling, the other is Coastguard—a sequence with more complex texture and dynamicity applies to advance presence.

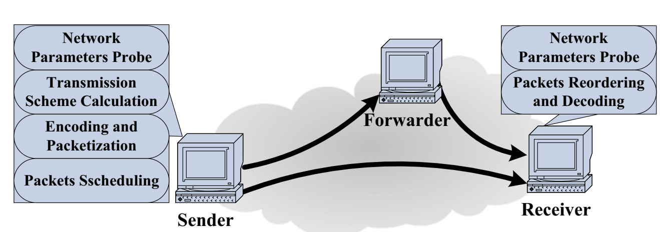

Fig. 6. Structure of online experiments.

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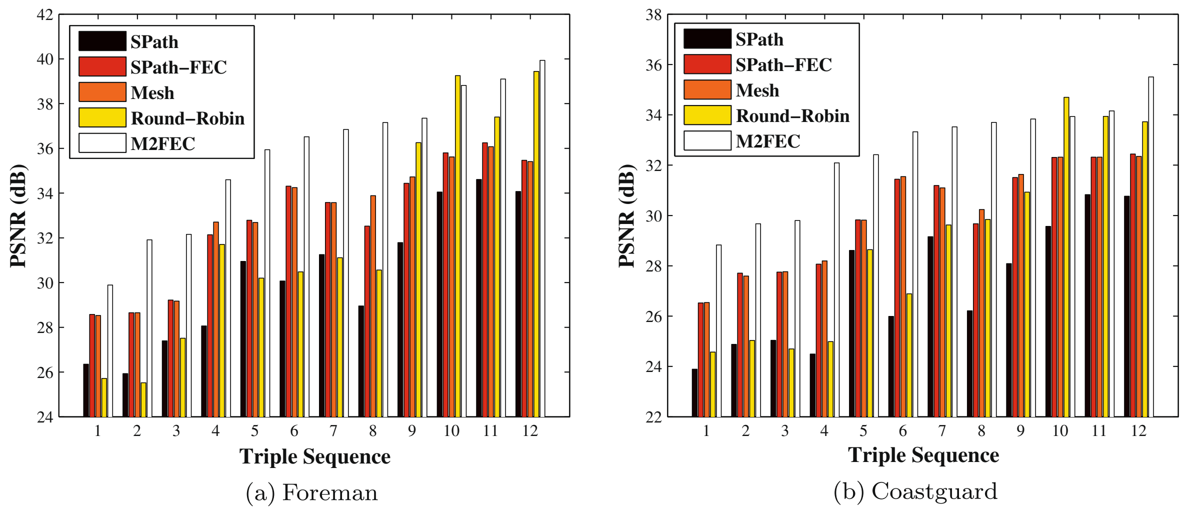


(a) Foreman (b) Coastguard

Fig. 7. PSNR bar of Short Term PlanetLab experiment.

Table 3
Average PSNR (dB) across the 12 PlanetLab Triples.

	SPath	SPath-FEC	Mesh	Round-Robin	M2FEC
Foreman	30.3	32.8	32.9	32.1	35.8
Coastgurad	27.3	30.1	30.1	29.0	32.6

There are two types of measurements in this section: one is Short Term, where each measurement lasts short time but is repeated several times at different period to filter out accidental factor; on contrast, the other is Long Term, where each measurement lasts long time to investigate the effects of Internet dynamics on every schemes.

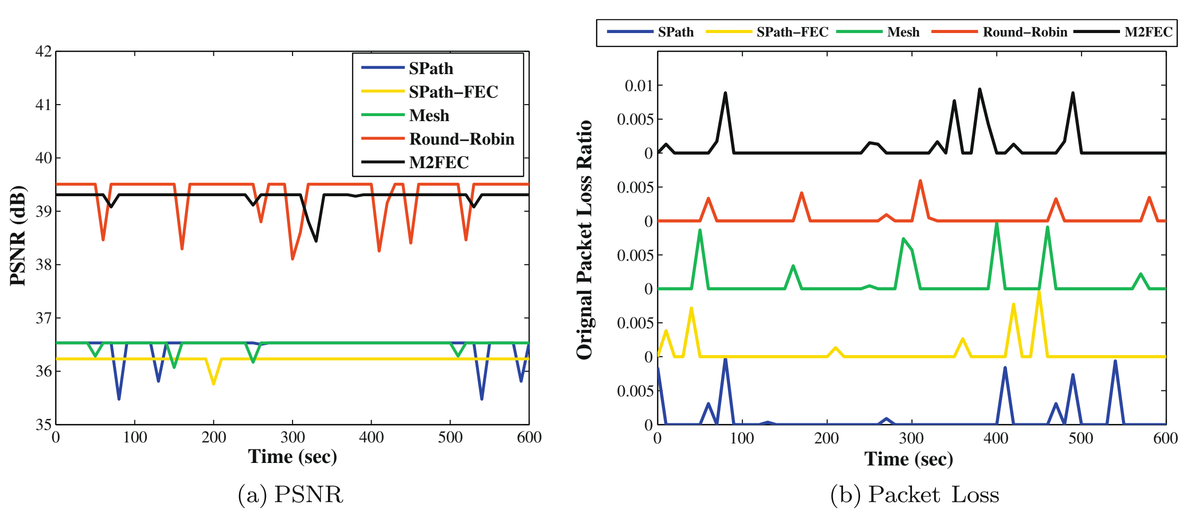
5.2. Short Term results

We define a group of sender, forwarder and receiver as a Triple. There are 12 Triples in Short Term measurement. In this experiment, each kind of transmission scheme lasts 1 min and repeats 10 times on one Triple to filter out accidental perturbation. The average PSNR across the 10 times measurement are summarized in Fig. 7(a) and Table 3. On average, M2FEC improves PSNR by 5.5 dB under Foreman sequence and 5.3 dB under Coastgurad sequence than SPath scheme.

It should be pointed out that there are a few special cases during the measurement. For example, Round-Robin performs 0.5 dB better than M2FEC in Triple 10, because there is almost no loss during the transmission and the redundancy in FEC coding decreases the aggregate source encoding bitrates which is a critical factor to media quality. In contrast, in Triple 1 and Triple 2 under Foreman sequence, Round-Robin performs rather poor, even worse than SPath, due to heavy loss along the overlay path.

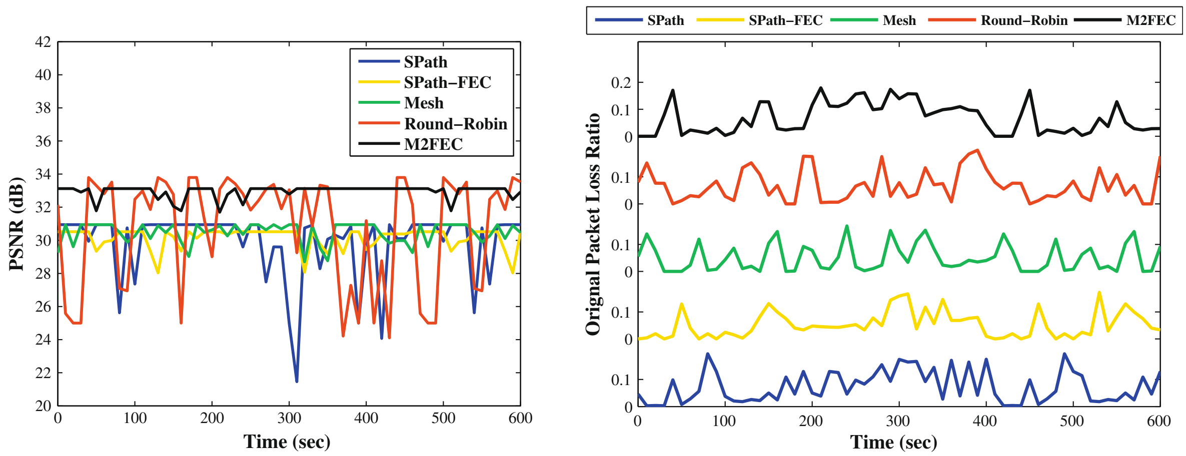
5.3. Long Term results

In the Long Term measurement, each transmission last 10 min and the same CIF sequence is resent 60 times consecutively (each sequence lasts 10 seconds long). Fig. 8(a) shows a situation with durative good network condition (on average 0.5% loss and 800 kbps bandwidth allocated along each path). In this scenario, M2FEC successfully avoid some of the sudden PSNR degradation which frequently happens in Round-Robin. On the contrary, Fig. 9(a) shows a situation with rather poor network condition (on average 7% loss and only 200 kbps bandwidth allocated along each path). Almost no profits can be obtained with Round-Robin



(a) PSNR (b) Packet Loss

Fig. 8. A Triple's fluctuation under good network condition (Foreman).



(a) PSNR (b) Packet Loss

Fig. 9. A Triple's fluctuation under poor network condition (Foreman).

as the its fierce PSNR variation, while in M2FEC and SPath-FEC, the variation can be successfully reduced by 80%.

It should be emphasized that our comparison is fair enough as we did not pick up good network time slot for FEC schemes while bad time slot for non-redundancy schemes. As shown in Figs. 8(b) and 9(b), during the measurement the original packet loss of each schemes does not exhibit notable difference.

6. Conclusion and future work

This paper proposes a framework for analyzing and comparing a variety of multi-path transmission strategies in quantity and develops M2FEC—a FEC based transmission scheme for interactive multimedia applications. M2FEC takes network conditions and application properties into consideration and optimizes media quality at the client by optimizing proposed model online. A series of theoretical analysis and actual Internet experiments validate the superiority and feasibility of M2FEC. Though only comparing with SPath, SPath-FEC, Mesh and Round-Robin in this paper, it is not difficult to extend out framework to other schemes such as MDC or LC based multi-path transmission.

It should be pointed out that media quality degradation happens in a few PlanetLab measurements even under M2FEC transmission. Several possible reasons may cause these degradations. First, FEC protection is not strong enough as network parameters measurement and media transmission are asynchronous. Secondly, heavy bursting packet loss leads to the invalidation of FEC coding. Several efforts could be made to improve current M2FEC. For example, adjust transmission parameters timely when network condition changes; employ more accurate network model on packet loss, delay distribution; introduce other error-resilient coding such as MDC or LC into our framework to further increase robustness and reliability. These mentioned solutions are also our valuable but challenging future work.

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Appendix A. A proof of packet loss when employed FEC

Assume the packet loss follows the Bernoulli distribution with parameter p. Furthermore, the packet losses are independent and identically distributed. For a Reed–Solomon code $RS(n,k)$ which contains k data packets and $n-k$ redundant packets in each FEC block, the irrecoverable loss probability is the probability that less than k packets are received per n packets. We define a random variable $Y (0 \leqslant Y \leqslant k)$ to be the number of data packets that could not be recovered in each FEC block. In Bernoulli model with the path packet loss rate of p, we have:

$$P[Y=0] = \sum_{i=k}^{n} \binom{n}{i} (1-p)^i p^{n-i}$$

$$P[Y=1] = \binom{k}{1} (1-p)^{k-1} p.p^{n-k}$$

$$P[Y=2] = \binom{k}{2} (1-p)^{k-2} p^2.\left[p^{n-k} + \binom{n-k}{1} p^{n-k-1}(1-p)\right]$$

$$\cdots$$

$$P[Y=k-1] = \binom{k}{k-1} (1-p) p^{k-1} \sum_{i=0}^{n-k} \binom{n-k}{i} p^{n-k-i}(1-p)^i$$

$$P[Y=k] = p^k \sum_{i=0}^{n-k} \binom{n-k}{i} p^{n-k-i}(1-p)^i$$

The expected number of irrecoverable packet in each FEC block can then be computed as:

$$\begin{aligned}
E[Y] &= \sum_{j=0}^{k} jP[Y=j] \\
&= kp.p^{n-k} + \left[kp - \binom{k}{1}(1-p)^{k-1}p\right]\binom{n-k}{1} p^{n-k-1}(1-p) \\
&\quad + \cdots + \left[kp - \sum_{i=1}^{n-k} i \cdot \binom{k}{i}(1-p)^{k-i}p^i\right]\binom{n-k}{n-k}(1-p)^{n-k} \\
&= kp - \left[\sum_{i=1}^{n-k} i \cdot \binom{k}{i}\binom{n-k}{i}\right](1-p)^k p^{n-k} - \\
&\quad - \left[\sum_{i=1}^{n-k-1} i \cdot \binom{k}{i}\binom{n-k}{i+1}\right](1-p)^{k+1}p^{n-k-1} \\
&\quad - \cdots - \binom{k}{1}\binom{n-k}{n-k}(1-p)^{n-1}p \\
&= kp - k\sum_{i=1}^{n-k}\binom{n-1}{n-k-1}(1-p)^{k+i-1}p^{n-k-i+1}
\end{aligned}$$

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Table 4
Candidate nodes in Planetlab experiment.

Domain Name	Location
planetlab2.cs.stevens-tech.edu	Eastern USA
planet1.cs.rochester.edu	Eastern USA
planetlab-1.cmcl.cs.cmu.edu	Eastern USA
planetlab01.cs.washington.edu	Eastern USA
server1.planetlab.iit-tech.net	Middle USA
planetlab1.csres.utexas.edu	Middle USA
planetlab1.flux.utah.edu	Western USA
planetlabnode-1.docomolabs-usa.com	Western USA
node-1.mcgillplanetlab.org	Canada
planetlab-1.it.uu.se	Sweden
planetlab-02.ece.uprm.edu	Puerto Rico
planetlab02.tkn.tu-berlin.de	Germany
planetlab2.cs.ucl.ac.uk	U.K.
planetlab2.eurecom.fr	France
planetlab2.ie.cuhk.edu.hk	Hongkong
planetlab1.iii.u-tokyo.ac.jp	Japan
xjtu1.6planetlab.edu.cn	China
thu2.6planetlab.edu.cn	China
plab2.nec-labs.com	USA
pli1-pa-4.hpl.hp.com	USA

Therefore, the irrecoverable packet error probability after $RS(n,k)$ is

$$p_F = \frac{1}{k}E[Y] = p - \sum_{i=1}^{n-k} \binom{n-1}{n-k-i} (1-p)^{k+i-1} p^{n-k-i+1}$$

Appendix B. Nodes in our testbed

The 20 nodes in our testbed are listed in Table 4.

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