Forward Error Correction Strategies for Media Streaming over Wireless Networks

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ABSTRACT

The success of next-generation mobile communication systems depends on the ability of service providers to engineer new added-value multimedia-rich services, which impose stringent constraints on the underlying delivery/transport architecture. The reliability of real-time services is essential for the viability of any such service offering. The sporadic packet loss typical of wireless channels can be addressed using appropriate techniques such as the widely used packet-level forward error correction. In designing channel-aware media streaming applications, two interrelated and challenging issues should be tackled: accuracy of characterizing channel fluctuations and effectiveness of application-level adaptation. The first challenge requires thorough insight into channel fluctuations and their manifestations at the application level, while the second concerns the way those fluctuations are interpreted and dealt with by adaptive mechanisms such as FEC. In this article we review the major issues that arise when designing a reliable media streaming system for wireless networks.

INTRODUCTION

The advance of wireless networks is turning huge research interest into tremendous commercial success. Wireless local area networks (WLANs) are the most prominent example. With the ever increasing capacity of such networks, many interactive services (collaborative IP tools, VoIP, video streaming, push services, etc.) are emerging. Thanks to the rise of powerful video compression techniques such as H.264 and MPEG-4, it is now possible to combine video, audio, and data within the same signal and transmit it over packet-based wireless networks. These advances enable the emergence of new value-added multimedia applications with significant business promise. For example, service providers (SPs) may further increase their customer base by expanding the range they provide of quality of service (QoS)-enabled video services over wireless networks.

Given these advances, various indoor and outdoor WLAN network operators are now more and more concerned about their ability to provide multimedia services with sustained QoS guarantees to a large number of heterogeneous wireless terminals. Providing QoS guarantees is a major imperative in developing viable business models, while serving a large number of users is an obvious business goal. Although limited terminal capabilities represent a handicap for mobile streaming systems, communications reliability remains the main issue when streaming media over WLANs. Error control techniques are required in such environments, where the packet loss process inherent to wireless channels is usually *bursty*. IEEE 802.11 link-layer retransmissions are efficient only on a shorter timescale and in the face of short-term fluctuations (fast fading); more persistent fluctuations (slow fading) render these mechanisms inefficient. Application-level error control techniques provide additional reliability on a longer timescale.

In this article we first focus on a better understanding of the WLAN *burstiness* effect in order to characterize the channel behavior by more accurate QoS metrics. We combine a loss-run length model and an inter-loss distance model to accurately capture both the channel burstiness and the spacing between loss runs. This combined loss model, operating at the multimedia server side, relies on accurate network feedback (extended Real-Time Control Protocol, RTCP) that indicates, for each transmitted packet, whether it was lost or received. The objective is to leverage loss-specific metrics to gain a better understanding of the loss process, and ultimately improve the responsiveness and efficiency of forward error correction (FEC). We then investigate the potential gain from using different FEC optimization techniques. In particular, interleaving can considerably enhance FEC recovery capabilities by dispersing the effects of correlated (clustered) packet loss, resulting in improved FEC recovery capabilities [1].

Clearly, it is important to integrate adaptive FEC techniques with both the service characteristics (video coding semantics/metadata) and the underlying transport layer. The objective of such cross-layer integration is to regulate error control aggressiveness according to:

- The video packets' relevance to end users' perceived quality
- The channel loss trends reported by the underlying transport layer

The remainder of this article is organized as follows. We introduce the key aspects and features of FEC techniques. Different FEC optimizations for efficient adaptation are presented. We present an accurate short-term channel characterization in terms of packet loss distribution to improve the effectiveness of FEC adaptation. We present some approaches to combine adaptive FEC optimizations with packet loss distribution modeling. Concluding remarks are then given.

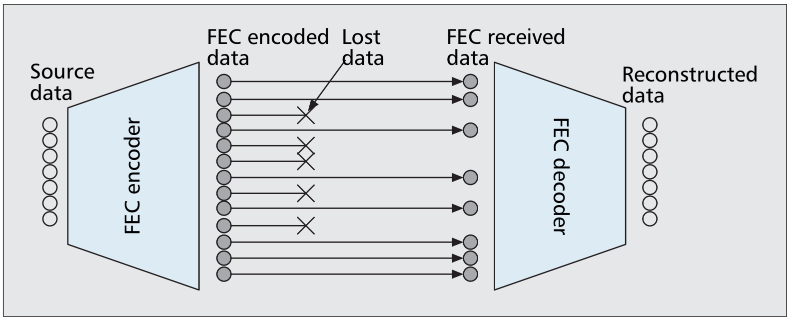


Figure 1. *Forward error correction encoding.*

Forward Error Correction Principles

FEC is a method commonly used to handle losses in real-time communication. FEC techniques enable a receiver to correct errors/losses without further interaction with the sender. An (n, k) block erasure code converts k source data into a group of n coded data, such that any k of the n encoded data can be used to reconstruct the original source data. Usually, the first k data in each group are identical to the original k source data; the remaining $(n - k)$ data are referred to as parity data (Fig. 1).

Usually, FEC codes are able to correct both errors and erasures in a block of n symbols. In coding theory an error is defined as a corrupted symbol in an *unknown* position, while an erasure is a corrupted symbol in a *known* position. In the case of streamed media packets, loss detection is performed based on the sequence numbers in RTP packets (i.e., *erasure* codes).

FEC Coding Techniques

In FEC codes redundant data are derived from the original data using techniques from coding theory. Traditional error correcting codes, such as parity, LDPC, Reed-Solomon, and Hamming codes, have recently attracted a substantial amount of attention. LDPC, XOR (exclusive-OR), and Reed Solomon codes are the most popular schemes proposed in the literature and are often recommended in Internet Engineering Task Force (IETF) Real-Time Transport Protocol (RTP) profiles. We briefly describe the two most widely used erasure codes; for details on other schemes the reader is referred to [2].

Reed-Solomon Codes — Reed-Solomon (RS) codes are a special class of linear non-binary block codes with the ability to correct both errors and erasures in a block of n symbols (packets). An RS code achieves ideal error protection against packet loss since it is a maximum distance separable (MDS) code, which means that no other coding scheme can recover lost source data symbols from fewer received code symbols. Besides the block size limitation, RS codes also suffer from computational complexity [3].

Low Density Parity Check Codes — Many recent research works have considered the use of low density parity check (LDPC) code. These codes have two main advantages:

- They use XOR operations for high-speed encoding/decoding, which are more suitable for handheld receiver devices.
- They operate on very large source blocks.

However, LDPC are not MDS codes, which means that they are less bandwidth-efficient than RS codes.

Recent research, carried out in the context of the IETF Multicast Reliable Transport (MRT) working group, performed evaluation of FEC codes and pointed out that LDPC codes are more suitable for large block transfers over unidirectional channels, while RS codes are more appropriate for small block size and real-time streams.

Byte-Level vs. Packet-Level FEC

FEC can be done at many levels from byte level up to packet level. In byte-level FEC, a symbol is a byte; while in packet-level FEC, a symbol is a packet. Byte-level FEC is implemented at the physical layer of almost all wireless networks. Packet-level FEC consists of producing h redundant packets from k original ones. An FEC packet is generally based on erasure coding and its usefulness is due to:

- A single parity packet can be used to correct different single-packet losses in a group of packets.
- Byte-level FEC is unable to recover a completely lost or delayed packet.
- When using byte-level FEC, a corrupted packet is already detected and discarded at the link layer with cyclic redundancy check (CRC), or at the transport layer with CHECKSUM, and so will not be available at the application level.

Even though most existing wireless access networks use integrated physical layer adaptive coding and modulation schemes (e.g., IEEE 802.16a uses variable-rate RS/convolutional coding [CC] schemes and variable-modulation scheme), packet-level FEC protocols are usually required at the application level. Wireless communication experiences both:

- Short-term fast fading and white Gaussian noise, which is addressed by the integrated physical layer coding

The key contribution is that important media packets may benefit from complete protection, while only part of less important packets are protected . This approach is particularly useful with video codecs that enable data partitioning.

- Long-term slow fading (e.g., when entering a tunnel), which is addressed by packet-level FEC encoding

These two levels of FEC encoding are fully complementary, with each level addressing a different problem. However, there is a need for additional packet-level FEC protection to increase the reliability of multimedia communications in a wireless context.

Packet-Level FEC Performance Metrics

We now review the performance criteria for packet-level erasure codes. There are many key erasure-code performance metrics that determine the suitability of each code. A thorough study of packet-level FEC performance [2] has identified the following metrics:

Redundancy ratio — The ratio n/k is usually referred to as the stretch factor of an erasure code. The stretch factor quantifies the amount of redundancy with respect to the source data.

Decoding inefficiency ratio (inef_ratio) — This represents the minimum number of packets required to recover an FEC block divided by the number of source packets. Typically, the *inef_ratio* is equal to one in MDS codes, while it is slightly higher in non-MDS codes. It is calculated as follows:

$$inef\_ratio = \frac{\text{Number of Packets Required for Decoding}}{\text{Number of source packets}}$$

Encoding/decoding times and bandwidth — Measuring the time (*time*) needed to encode/decode an FEC block of a certain erasure codes class (RS, LDPC, etc.) is useful for computing the achievable bandwidth in a real-time streaming system and the suitability of such codes for resource-constrained wireless terminals. This bandwidth is calculated as follows:

$$BW = \frac{n \cdot \text{pkt size (in bits)}}{\text{time}}$$

Due to the complexity of error correcting codes, the processing capacity of both the source and destination should also be considered in designing FEC-based reliable communication systems.

FEC Code Optimizations

Unequal FEC Protection

Unequal error protection is a powerful technique that may considerably improve the end user's perceived quality. Based on knowledge of content priority and/or video stream framing, such techniques appropriately distribute the redundancy budget over the video stream to reduce the impact of packet loss on the video quality perceived by users [4]. Authors in [5] propose the protection of MPEG-4 video objects (VOs) according to their priority and importance in the video scene. Nevertheless, it is admitted that, in certain circumstances, providing higher protection to the more important parts (e.g., Intracoded VOP) of a low-priority VO is better than protecting the less important parts (e.g., B-VOP) of a high-priority VO. Thus, depending on the protection strategy (favoring an important object or favoring the overall quality), it is possible to distribute the FEC budget in a way that best meets predefined objectives.

Within the IETF AVT working group, extensive ongoing discussions have led to the design of a generic RTP payload format [6] that provides integrated protection of media packets. This approach is based on the XOR (parity) operation. It allows end systems to apply protection using different protection lengths and levels. It also enables complete or partial recovery of the critical payload and RTP header fields, depending on the packet loss situation. The key contribution is that important media packets may benefit from complete (full length) protection, while only part of less important packets are protected (RTP header and a certain number of bytes). This approach is particularly useful with video codecs (e.g., H.264) that enable data partitioning.

Interleaved FEC Protection

Data interleaving is commonly used in video streaming systems to reduce the effects of loss. The sender re-sequences the packets before transmitting them to the receiver, so that originally adjacent packets are separated by a distance that may vary over time. The interleaving disperses the effect of packet losses and mitigates the effect of bursty losses on multimedia decoding. The key advantage of interleaving is that it provides better error resiliency while not increasing the bandwidth requirement of a stream. It is particularly effective for multimedia streams with short-term dependencies between data (e.g., predictive spatial/temporal coding); here, adjacent losses result in error propagation, which affects the quality of decoded content as even correctly received data might not be decoded properly. However, the pitfall of interleaving consists in the fact that it increases latency due to additional buffering at the sender. This limits the application of this technique to delay-sensitive interactive applications.

Interleaved FEC protection is based on the combination of two well-known techniques, FEC and interleaving. This combination may increase FEC efficiency and consequently reduce the amount of FEC transmission at servers. FEC techniques are not sufficient to safeguard data transmissions from burst errors, as they are only effective in counteracting random losses. FEC interleaving is capable of minimizing the effect of burst errors at the decoding level, although its efficiency still depends on the amount of FEC redundancy being transmitted and the interleaving stride used.

In order to reduce the effect of clustered losses, redundant information could be added into temporally distant packets [1], which introduce even higher delay. Hence, the repair capability of FEC is mainly limited by the delay budget. Reference [1] proposed an adaptive FEC based on transmission of redundant audio samples (audio packets), so information relative to packet n may be spread over multiple packets, and relying on a simple two-state Gilbert model to react to network fluctuations. This scheme is efficient for Internet telephony, since a Gilbert model suffices to capture the rather slow loss dynamics.

It is worth mentioning that excessive interleaving may lead to uncertain FEC efficiency improvement, depending on the loss pattern exhibited by the wireless link. Clearly, it is important to capture the wireless channel dynamics through pertinent QoS metrics in order to determine the appropriate interleaving level to maximize the effectiveness of FEC recovery.

FEC Redundancy Control

Typical packet-level FEC protocols that use k media packets to produce n packets, including $h = n - k$ parity packets, have the capacity to overcome up to h packet losses (when using MDS codes). This provides resiliency against a maximum packet loss rate of $p = h/n$ when considering that even FEC packets may be affected by loss. Thus, based on averaged packet loss rate (p) measurements such as that provided by RTCP feedback, it is possible to constantly adjust the redundancy amount h as follows:

$$h = \frac{p \cdot k}{(1-p)}$$

Many approaches use this simple model with varying levels of complexity when dealing with application-level framing (e.g., video frame fragmentation). The maximum acceptable loss rate threshold beyond which the streaming server triggers FEC adaptation may differ depending on the nature of the audiovisual content and its loss resiliency characteristics. This FEC adaptation model poses a problem when dealing with channels that exhibit varying packet loss rates over time. The frequency with which the network loss rate is reported to the sender may reduce the responsiveness of FEC schemes, leading to suboptimal FEC efficiency. A high frequency would enhance the responsiveness at the sender, while causing high variations between successive measurements and possibly leading to instability, not to mention excessive feedback traffic overhead. On the other hand, a low frequency would have good stability and low overhead but poor responsiveness.

FEC redundancy control is effective only in cases where packet losses are uniformly distributed over the transmission. In reality, the packet loss process is often variable over time, so the use of the average loss rate to adjust the FEC redundancy may not produce optimal FEC transmission. Obviously, the averaged loss rate is more useful for a very large FEC block, but fails to capture short-term loss process fluctuations. To address this issue, more advanced loss characterization models have been used to capture the correlation between packet losses [1], where it was shown that a simple Gilbert model can considerably improve the performance of an adaptive FEC system over the Internet by using the probability of the loss state, instead of the averaged loss rate, to control redundancy transmission.

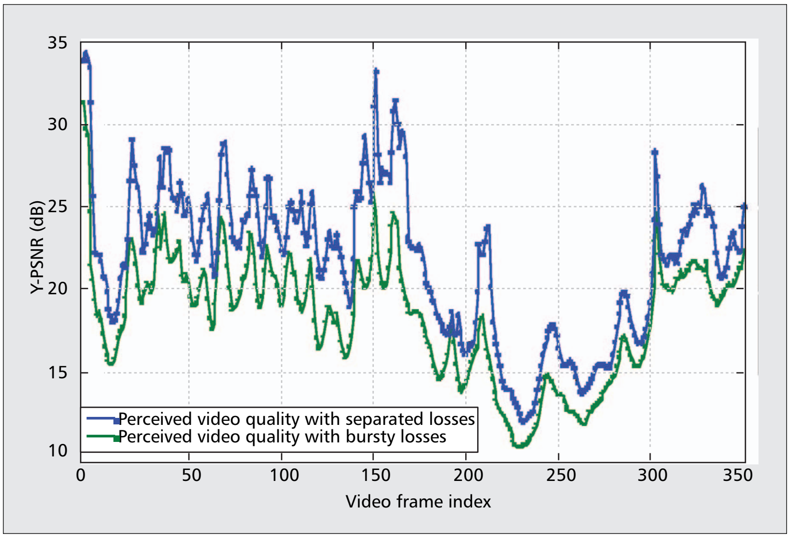


Figure 2. *Loss pattern effect on video quality.*

Short-Term Packet Loss Management

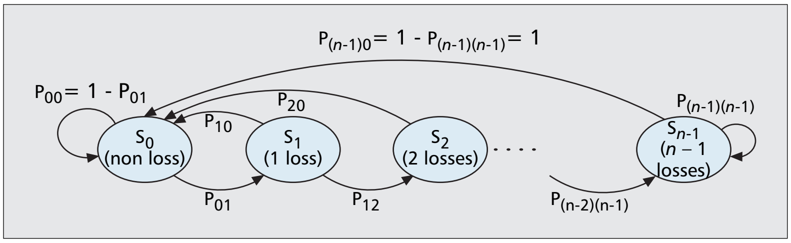
Impact of Loss Distribution on Streaming Media Quality

Typical WLAN communications are subject to high bit error rates, which usually occur through correlated (adjacent) packet losses. In certain real-time applications, the loss distribution (i.e., loss pattern) is a key parameter that determines the performance perceived by the users. Peak signal-to-noise ratio (PSNR) is often used to measure the objective quality of a reconstructed video stream. Figure 2 plots two PSNR curves corresponding to two specific loss patterns affecting the same video communication. Both loss patterns correspond to a loss percentage of 2 percent (30 lost packets over 1500 transmitted packets). In the first loss pattern the losses are well separated, while in the second loss pattern the loss events occur in bursts of three packets. In this experiment we used a Quarter Common Intermediate Format (QCIF — 176 × 144 pixels) Foreman H.264-coded sequence with an Intra period of 10.

Well separated losses lead to smooth video quality degradation due to the nature of the coded video bitstream, which contains a substantial amount of temporal and spatial dependencies. On the other hand, clustered losses decrease the decoding quality and considerably reduce the efficiency of H.264-integrated resiliency features. FEC performance is also significantly degraded when packet losses occur in bursts. It is clear that video quality depends on the specific loss pattern as well as the averaged loss rate.

Packet Loss Process Characterization

The high packet loss rates typical of WLAN communications are mainly due to channel fluctuations that cannot be fully addressed at the physical layer (slow fading). Thus, it is crucial to capture the channel dynamics at the packet level to effectively overcome packet loss. The packet



■ **Figure 3.** *Extended Gilbert model with limited states.*

loss process can exhibit high temporal dependency: for instance, if packet n is lost, packet $n + 1$ is likely to be lost as well. This introduces burstiness in network losses, and the performance of FEC efficiency is affected, resulting in poor packet loss recovery capabilities.

Several researchers have attempted to characterize the loss process by highlighting the correlation between loss and delays. This is particularly useful in preventing burst packet loss associated with congestion. However, delay measurement is not very revealing in a single-hop wireless communication scenario, where the loss event follows a random process due to location- and time-varying channel conditions; that is, loss events are not necessarily correlated with the available bandwidth. In the following we introduce basic mathematical concepts on which packet loss modeling is built.

Gilbert Model — In [7] a Markov model is proposed to capture temporal loss dependency. The work in [7] analyzed the two-state Markov model, also known as the Gilbert model. p denotes the probability that the next packet is lost, provided the previous one has arrived; q is the opposite. $(1 – q)$ is the conditional loss probability (*clp*). Typically, $p + q < 1$. If $p + q = 1$, the Gilbert model will have the Bernoulli model properties.

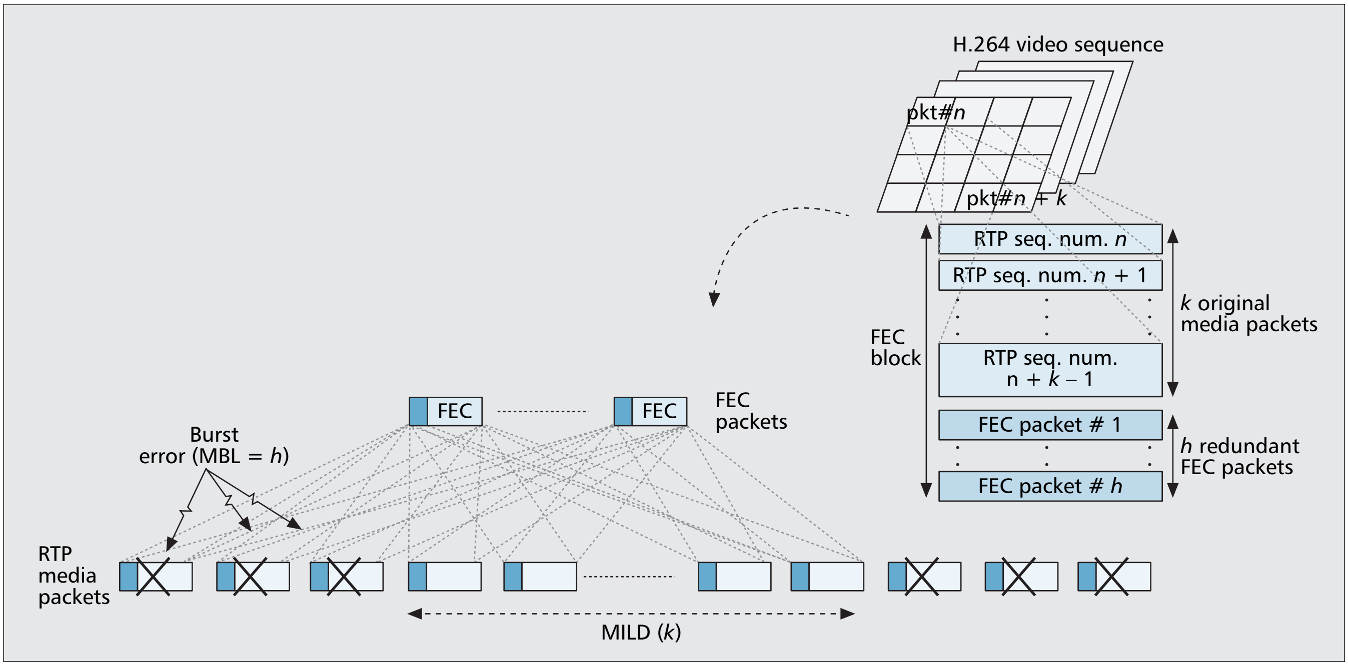
From the above definition, we can compute p_0 and p_1, the state probability for state 0 and 1, respectively. In the Gilbert model they also represent the mean arrival and loss probability, respectively. Note that many FEC redundancy control protocols (e.g., the one in [6]) use the probability of state 1 (p_1) to adjust the amount of FEC transmission.

$$p_0 = \frac{q}{p+q} \qquad p_1 = \frac{p}{p+q}$$

P_k (the probability distribution of loss runs of length k, i.e., k consecutive losses) has a geometric distribution.

Extended Gilbert Model — An extension of the Gilbert model has been used in modeling Internet losses [7]. It handles long loss runs by using models with multiple states. We consider using this model to capture the typical loss run length and typical distance between two successive loss runs. We define the random variable X as follows: $X = 0$: no packet loss; $X = k$: exactly k consecutive packets are lost; and $X \geq k$: at least k consecutive packets are lost. With this definition, we establish a loss run length (loss burst length) model with n states as shown in Fig. 3. We rely on extended RTCP feedback that continuously reports the measured loss pattern. Each RTCP report corresponds to the last 300 transmitted packets (i.e., loss pattern segment).

The system keeps a counter l, which is the number of consecutively lost packets. It is reset whenever the next packet is successfully delivered. The parameter to be determined is $P[X_i | X_{i-1}$ to X_{i-l} are all lost]. Let m_i for ($i = 1, 2 \ldots n – 1$) denote the number of loss bursts having length i, where $(n – 1)$ is the longest loss burst. m_0 denotes the number of delivered packets; n represents the number of states in the model. Note that a model can be completely character-



■ **Figure 4.** *Model-based FEC block allocation scheme.*

ized by its burst loss length occurrences vector M (i.e., the coefficient vector $M = (m_0, m_1, \dots m_{n-1})$). The mean burst loss length (MBL) is easily deduced and gives the expected mean loss run length based on the previously observed loss distribution:

$$MBL = \frac{\left(\sum_{i=1}^{n-1} i \cdot m_i\right)}{\sum_{i=1}^{n-1} m_i} \quad (1)$$

The interloss distance (ILD) metric was recently proposed within the IETF working group to describe the distance between packet loss events in terms of sequence number. The ILD metric is useful for complementing the loss model for enhanced loss pattern prediction and multimedia application adaptation in two respects: while an accurate loss model is able to model loss run distributions, it does not model distances between loss runs; and small ILD values may also degrade the performance of FEC codes.

As with the loss model, we derive a model to characterize ILD distribution. This is useful to understand and predict the spacing between loss events. Let d_i, $i = 1, \dots n - 1$ denote the number of ILDs having length i. The ILD model is completely characterized by its occurrence vector $D = (d_1, d_2, \dots d_{n-1})$. The mean ILD (MILD) is given by

$$MILD = \frac{\left(\sum_{i=1}^{n-1} i \cdot d_i\right)}{\sum_{i=1}^{n-1} d_i} \quad (2)$$

Loss-Pattern-Aware FEC Redundancy Allocation

In this work we consider an RTP/UDP-based streaming system, although TCP-based streaming systems are today by far the most dominant alternative in unmanaged/unreliable networks such as the Internet. RTP/UDP streaming systems are much more bandwidth-efficient and are considered more for deployment in managed networks such as broadband operators' networks.

To better understand the meaning of the MBL and MILD metrics, Fig. 4 illustrates their potential impact on adaptive FEC transmission. Here, adaptive FEC-capable communication is affected by two loss bursts having a length of MBL = 3. The two loss bursts are separated by k (MILD) correctly received packets. The FEC scheme is a video-frame-based FEC allocation, in the sense that packets stemming from each video frame will fit into one FEC block.

In conventional FEC transmitting h FEC packets in an FEC block of k source data provides an erasure resiliency against a packet loss rate of $h/(h + k)$. However, $h/(h + k)$ represents the averaged loss rate and does not give any indication of channel burstiness (i.e., loss clustering trends). It is better to use more specific loss distribution metrics such as MBL and MILD to help protect video packets. In fact, capturing transient wireless link burstiness may improve the responsiveness of the streaming server to short-term variations in network conditions.

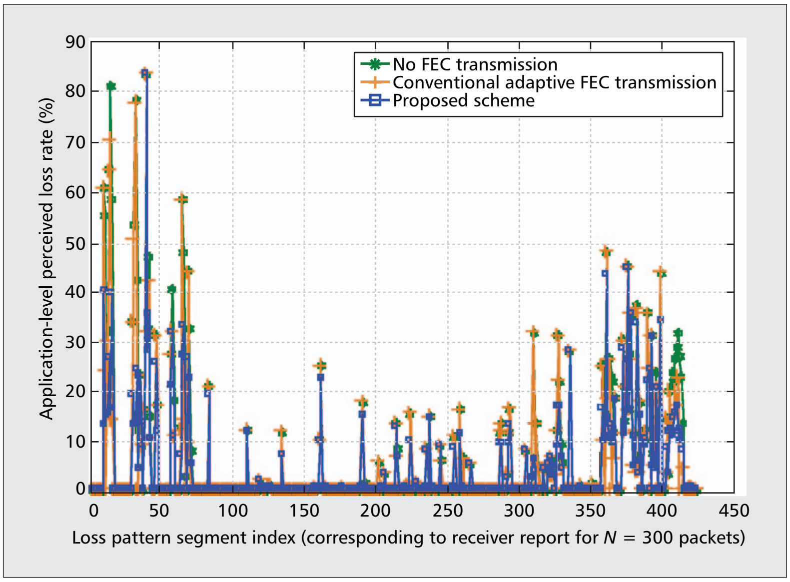


Figure 5. *Instantaneous receiver-perceived loss rate (i.e., after FEC recovery).*

Applying an appropriate amount of redundancy (h = MBL) makes the communication robust against the most likely expected packet loss run length. On the other hand, taking a number of original data k equal to MILD (k = MILD) improves the bandwidth efficiency of FEC codes. Since MILD represents the most likely spacing between two loss bursts, using it as k in FEC blocks reduces the probability of having two loss bursts affect the same FEC block.

We now compare conventional adaptive FEC with our loss-pattern-based adaptive FEC through trace-based simulation. This evaluation process is usually used to estimate the overall performance of loss models with respect to multimedia service quality. We extend this evaluation process with finer-grained measurements by estimating the model performance several times during the evaluation process. Trace-based simulation uses loss patterns collected from real traffic transmissions over a network. In our case we use a WLAN network to collect the loss patterns resulting from actual H.264 video multicasting (H.264 baseline profile at a mean bitrate of 466.7 kb/s.). Each transmitted RTP packet contains a sequence number intended for intramedia synchronization. If the packet arrives, the receiver writes the sequence number into the trace file. Afterward, during offline analysis, we calculate the final loss pattern and then divide it into several segments corresponding to RTCP reports. The simulation of FEC-enabled streaming server behavior is obtained after multipass processing. The number of processing passes strongly hinges on the number of loss pattern segments (RTCP reports).

Figure 5 gives the final loss rate values (after recovering with FEC) averaged over N = 300 packets, which represents the number of packets

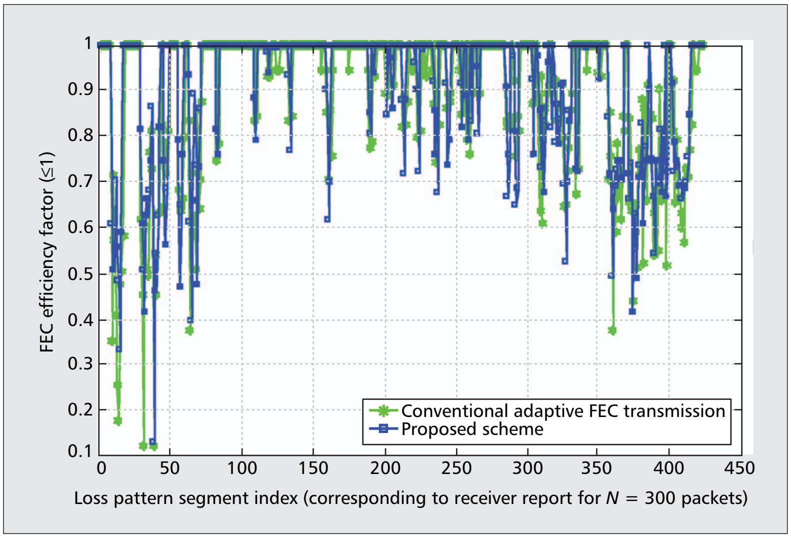


Figure 6. *Instantaneous measured FEC-EF.*

covered by a single feedback report. Our loss-model-based FEC adaptation is compared to conventional adaptive FEC. As described before, conventional adaptive FEC uses the total averaged loss rate p to adjust the amount of redundancy to be transmitted ($h = p \cdot n$).

The mean measured loss rate during the multicast streaming session is around 6.1 percent (7607 lost packets). Using the conventional adaptive streaming server, the mean perceived loss rate reached about 5 percent (6210 lost packets after FEC recovery). The perceived loss rate was about 4 percent (4968 lost packets after FEC recovery) with our loss-model-based FEC adaptation scheme. This difference in the mean loss rate could have devastating consequences at the video decoding level.

Our loss-model-based FEC usually requires more bandwidth than conventional adaptive FEC. The mean bandwidth consumption achieved with the conventional streaming system is around 495 kb/s, while the bandwidth consumption is about 510.5 kb/s with our proposed scheme. This excess bandwidth usage (3.1 percent) could be afforded by SPs if the video quality is sustained. Indeed, it is commonly accepted, from an SP's point of view, that additional bandwidth requirement is acceptable for the sake of enhanced loss resiliency and better perceived quality.

It is important to measure how accurate the FEC transmission is, that is, the percentage of FEC packets that are actually used to recover packet loss at FEC blocks decoding level; this excludes the FEC packets sent unnecessarily or those that are not sufficient to recover a given FEC block. For the purpose of measuring this, we define the FEC efficiency factor as

$$FEC - EF = \frac{\text{Total Data}}{\text{Received Data} + \text{Transmitted FEC}}$$

Here the total data represents the video packets obtained after recovering with FEC (i.e., received packets and recovered packets), while received data is the number of packets correctly received. So ideally, FEC-EF = 1 when

- No FEC is transmitted and the communication does not suffer from loss.
- All transmitted FEC redundancy is used to recover lost packets.

Figure 6 illustrates the measured FEC-EF throughout the multicast streaming session. The mean measured FEC-EF forconventional adaptive FEC is around 0.8874, while the mean measured FEC-EF is 0.91 with our proposed scheme. The FEC efficiency is particularly noticeable when the loss pattern segments are correlated and show certain cyclical behavior in terms of packet loss distribution (i.e., stable MBL and MILD measurements).

Conclusions

QoS provisioning for audiovisual services is a challenging task to make WLANs viable in a network operator's commercial offerings, and ultimately more attractive for widespread use. In this context multimedia service reliability is a prerequisite in any service level agreement between the service provider and its end users. To achieve the necessary level of service reliability, it is essential to design and deploy integrated error control techniques. In this article we present an adaptive packet-level FEC protocol meant to complement existing medium access control/physical layer short-timescale error control mechanisms. We investigate how the network conditions could be better characterized by using packet loss patterns as feedback for FEC adaptation. We derive new loss-specific metrics to better balance FEC aggressiveness and resource requirements, to achieve both adaptation accuracy and bandwidth efficiency.

References

[1] J-C. Bolot *et al.*, "Adaptive FEC-based Error Control for Internet Telephony," *Proc. IEEE INFOCOM '99*, New York, NY, Mar. 1999.

[2] V. Roca and C. Neumann, "Design, Evaluation and Comparison of Four Large Block FEC Codecs, LDPC, LDGM, LDGM staircase and LDGM Triangle, plus a Reed-Solomon Small Block FEC Codec," INRIA res. rep. RR-5225, June 2004.

[3] L. Rizzo, "Effective Erasure Codes for Reliable Computer Communication Protocols," *ACM Comp. Commun. Rev.*, vol. 27, no. 2, Apr. 1997, pp. 24–36.

[4] Y. Shan and A. Zakhor, "Cross Layer Techniques for Adaptive Video Streaming over Wireless Networks," *IEEE ICME '02*, vol. 1, Lausanne, Switzerland, Aug. 2002, pp. 277–80.

[5] T. Ahmed *et al.*, "Cognitive Video Streaming Over Next Generation Networks: A Cross-layer Approach," *IEEE JSAC*, vol. 23, no. 2, Feb. 2005, pp. 385–401.

[6] A. Li *et al.*, "RTP Payload Format for Generic Forward Error Correction," IETF Internet draft draft-ietf-avt-ulp-23.txt, Aug. 2, 2007, work in progress.

[7] H. Sanneck and G. Carle, "A Framework Model for Packet Loss Metrics based on Loss Run Length," *Proc. SPIE/ACM SIGMM MMCN*, Jan. 2000.

Additional Reading

[1] A. Nafaa and A. Mehaoua, "Joint Loss Pattern Characterization and Unequal Interleaved FEC Protection for Robust H.264 Video Distribution over Wireless LAN," *Comp. Networks J.*, vol. 49, no. 6, Dec. 2005, pp. 766–86.

[2] K. Park and W. Wang, "AFEC: An Adaptive Forward Error Correction Protocol for End-to-End Transport of Real-Time Traffic," *Proc. IEEE IC3N*, San Diego, CA, Oct. 1998.

[3] J. Lu *et al.*, "Progressive Source-Channel Coding of Images Over Bursty Error Channels," *Proc. IEEE ICIP '98*, Chicago, IL, Oct. 1998.

BIOGRAPHIES

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