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Online AL-FEC policy problem on mobile unicast services

C. Bouras *, N. Kanakis

Computer Engineering and Informatics Department, University of Patras, Patras, GR 26504, Greece
Computer Technology Institute and Press "Diophantus", N. Kazantzaki Str., Patras, GR 26504, Greece

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ABSTRACT

Forward error correction (FEC) is a method for error control of data transmission adopted in several mobile multicast standards. FEC is a feedback free error recovery method where the sender introduces redundant data in advance with the source data enabling the recipient to recover from different arbitrary packet losses. Recently, the adoption of FEC error control method has been boosted by the introduction of powerful Application Layer FEC (AL-FEC) codes, e.g. RaptorQ codes. Furthermore, several works have emerged aiming to address the shortcomings of AL-FEC protection application utilizing deterministic or randomized online algorithms to enhance the efficiency of AL-FEC error control method. In this work, since the investigation of AL-FEC application as primary or auxiliary error protection method over mobile multicast environments is a well investigated field but the opportunity of utilizing the AL-FEC over mobile unicast services as the only method for error control replacing common feedback based methods that are now considered to be obsolete, we provide an analysis on the feasibility of AL-FEC protection over unicast delivery utilizing online algorithms in conjunction with AL-FEC codes with exceptional recovery performance.

1. Introduction

Forward error correction (FEC) is a method for error control of data transmission adopted in several mobile multicast standards. In multicast delivery, the FEC encoding significantly reduces the effect of independent losses at different receivers, while achieving a reduction in the rate of packet loss according to the introduced redundancy by the FEC encoder, resulting in large mitigation to the costly need of lost packets retransmission. Based on the above, several mobile multicast standards [1,2] recommend the use of FEC on application layer, and more specifically, Raptor codes family [3] are adopted due to their high performance. However, FEC protection comes with its own cost since controlling the introduced redundancy is not a trivial issue. The multicast sender should decide on the redundancy will introduce to the transmission so as to ensure that the multicast recipients will be able to recover independent data losses while, at the same time the redundant information should be adapted to the current reception conditions to avoid resources wastage. Based on this, the efficient application of AL-FEC protection can be achieved by a multicast transmitter enabled to adapt the introduced AL-FEC redundancy according to the current reception conditions. The design of an algorithm adapting the introduced AL-FEC transmission overhead can be reduced in the basis of an online problem [4]. Online problems assume that complete knowledge of the entire input is not available to an algorithm and the input is revealed in parts, with an online algorithm responding to each new input upon arrival.

In general, online algorithms [5] are used to confront problems where the input of the algorithm is not available in advance. Subsequently, online algorithms have to generate output without knowledge of the entire input since input information arrives in the future and is not accessible at present. In some problems, where the application of deterministic solutions lacks of applicability, a randomized online algorithm [4] is the simplest available algorithm and some times the most efficient solution. The effectiveness of online algorithms is evaluated using competitive analysis. The main concept of competitiveness is to compare the output generated by an online algorithm to the output produced by an optimal offline algorithm which knows the entire request sequence in advance and can serve it with minimum cost. The competitive ratio of an online algorithm A is defined with respect to an adversary. In general, the adversary generates a sequence σ and the online algorithm A has to serve σ. When constructing the sequence σ, the adversary always knows the description of the online algorithm A. Formally, given a sequence σ, $A(\sigma)$ denotes the cost of the online algorithm A and $OPT(\sigma)$ denotes the cost of the optimal offline algorithm. An

* Corresponding author at: Computer Engineering and Informatics Department, University of Patras, Patras, GR 26504, Greece.
E-mail addresses: bouras@cti.gr (C. Bouras), kanakisn@cti.gr (N. Kanakis).

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online algorithm A is called c-competitive if there exists a constant α such that $A(\sigma) - c \cdot OPT(\sigma) \leq \alpha$.

The scope of this work is to examine the feasibility of utilizing online AL-FEC protection as the primary error control method in evolved unicast mobile services based on the online AL-FEC policy problem. The proposed algorithms on the online AL-FEC policy problem have been evaluated so far only on multicast environments. However, the problem statement and the design of those online algorithms provide solutions that are not coupled with multicast environments and can be directly applied on unicast transmission. Furthermore, the option of utilizing AL-FEC protection as the primary error control scheme is boosted by the emergence of the powerful RaptorQ FEC codes that came to mitigate the major drawback of the predecessor Raptor codes i.e., the practically zero overhead.

2. Related work

Online algorithms are widely utilized in many research fields of mobile networks over several perspectives. The work presented in [6] proposes a data selection policy where, in the concept of competitive analysis, the decision of transmitting source data, retransmitting a packet or transmitting a redundant codeword is investigated. The authors provide a theoretical network model under which they design an online algorithm on choosing what data the multicast source should place in each sent packet. Furthermore, they provide trace-driven simulations to verify the effectiveness of the proposed scheme. The work presented in [7], examines the frequency assignment problem introducing distributed online algorithms. The examined online problem is abstracted as a multicoloring problem on a weighted graph and the authors propose a series of online algorithms on this basis. In the context of energy constraints and the design of routing algorithms, the authors of [8] propose an online algorithm on maximizing the throughput of multihop radio networks. In [9] online algorithms are utilized on multicast routing problems over energy-constrained ad-hoc networks. The authors propose two online algorithms on maximizing the capacity and lifetime of ad-hoc wireless networks and provide simulation results investigating the performance of the online algorithms. The work presented in [10] introduces a competitive online algorithm in terms of energy efficiency and delay in scheduling problems over wireless multicast environments. By reducing the energy-efficient transmission scheduling problem to a convex optimization problem, the authors design a variety of online algorithms aiming to minimize the energy required to transmit packets in a wireless environment. Furthermore, the authors of [11] present a set of randomized online algorithms studying the maximum independent set problem in disk graphs which can model resource allocation problems in mobile networks.

The authors of this manuscript introduced in [12] an online framework for the utilization of online algorithms on the efficient application of AL-FEC protection problem over mobile multicast networks evaluating the first attempt of a naive randomized online algorithm for the stated AL-FEC policy online problem. Moreover, the same authors presented in [13] a deterministic online algorithm based on weights assignment in each AL-FEC processed packet adapting the introduced AL-FEC overhead according to some encoding properties of RaptorQ AL-FEC code. Finally, the authors of this work presented in [14] an initial investigation of online AL-FEC application algorithms over mobile unicast services.

3. AL-FEC policy online problem

3.1. Unicast mobile services

The 3GPP packet-switched streaming service (PSS) is a standard for audio and video streaming to mobile terminals and provides a complete streaming and download framework for commercial content. The main scope of PSS is to define an application providing synchronized streaming of timed media, such as speech/audio, video, and text. 3GPP PSS is mainly based on protocols developed by IETF. The main protocols include the Real-Time Streaming Protocol (RTSP) for session control, the Session Description Protocol (SDP) for presentation descriptions, and the Real-time Transport Protocol (RTP) for media transport. In addition, HTTP is used for download of scene and presentation descriptions. SDP is used to describe a session offered by a PSS streaming server. Such a description is not subject to negotiation, although a client has the option to accept or reject it. However, in order to furnish a presentation with a high likelihood of acceptance, the server should consult the capability profile of the client, if provided. The main protocol controlling the streaming session is RTSP: all streaming flows are controlled by RTSP requests from the client to the server, where the media is streamed using RTP/UDP/IP. Moreover, 3GPP PSS also recommends the implementation of RTP retransmissions that enable repairs due to packet losses. RTSP [15] may use either an unreliable datagram protocol (UDP), a reliable datagram protocol (RDP) or a reliable stream protocol such as TCP as it implements application-level reliability. Requests are acknowledged by the receiver unless they are sent to a multicast group. If there is no acknowledgment, the sender may resend the same message after a timeout of one round-trip time (RTT) and it is implementation dependent. If a reliable transport protocol is used to carry RTSP, requests must not be retransmitted and the RTSP application must instead rely on the underlying transport to provide reliability. If both the underlying reliable transport such as TCP and the RTSP application retransmit requests, it is possible that each packet loss results in two retransmissions. The receiver cannot typically take advantage of the application-layer retransmission since the transport stack will not deliver the application-layer retransmission before the first attempt has reached the receiver. If the packet loss is caused by congestion, multiple retransmissions at different layers will exacerbate the congestion.

Retransmission of lost packets is an obvious mean by which losses can be repaired. However, it is typical that in some applications, this error control method cannot always perform well. In addition to the possibly high latency, there is a high bandwidth overhead introduced to the use of retransmission. Not only are the same data sent multiple times, but additional control traffic is necessary to realize the request for the retransmission. It has been shown that, under certain circumstances, the overhead of requesting retransmission for most packets may be such that the use of a FEC is more acceptable and efficient. This leads to a natural synergy between the two mechanisms, with a forward error correction scheme being used to repair all single packet losses, and those receivers experiencing burst losses, and willing to accept the additional latency, using retransmission based repair as an additional recovery mechanism. Similar mechanisms have been used in a number of reliable schemes.

3.2. RaptorQ codes

The outcome of the progress on erasure codes is the emergence of an enhanced Raptor code at Internet Engineering Task Force (IETF) [16] in order to address the drawbacks of the standardized Raptor code. The newer member in Raptor codes family is known as RaptorQ code. Raptor codes are fountain codes, meaning that the AL-FEC encoder can generate as many encoding symbols as desired on-the-fly from the source symbols of a source block of data. The decoder is able to recover the whole source block from any set of AL-FEC encoding symbols only slightly more in number than the number of source symbols. Raptor codes are systematic fountain codes producing n encoding symbols from $k < n$ source symbols. RaptorQ is a significantly more efficient AL-FEC code than its predecessor Raptor code, in terms of superior flexibility and higher protection and coding efficiency. The encoding process of RaptorQ code is mostly identical with that of Raptor code. However, RaptorQ code introduces certain design selections that ensure superior performance compared with that of Raptor code. A key differentiation between the two schemes is that the standardized Raptor code operates over Galois field GF(2) [17], while the enhanced RaptorQ code uses symbol operations over GF(256) [16] instead of over GF(2). Operating over larger finite fields allows RaptorQ to overcome the performance limitations of Raptor code since utilizing larger finite fields offers the potential of achieving recovery with lower reception overhead than the existing Raptor code. Moreover, additional important aspects of the enhanced properties of RaptorQ code are the increased number of possible source symbols and the increased number of generated encoding symbols. More precisely, RaptorQ can encode up to 56 403 source symbols into a source block in contrast to 8192 of the Raptor code and furthermore can generate up to 16 777 216 encoding symbols, 256 times more than the older Raptor code. The expanded range of these two parameters simplifies the application of the AL-FEC protection and offers higher flexibility to RaptorQ. Based on the properties of RaptorQ code, it is obvious that can perform better and more flexible both for file delivery and streaming services. Since RaptorQ can deliver files up to 3.4 GB as a single source block maximizes the decoding efficiency and protection due to the spreading of protection across the whole file, particularly for very large files. On the delay-sensitive real-time applications, the flexible range of the block size parameter allows to determine a QoS trade-off between protection and latency considering the delay constraints of the transmitted application. At the same time RaptorQ achieves lower computational complexity [18] than the older Raptor code.

Concerning the performance of RaptorQ, as already mentioned, the key property of a Raptor codes member is the probability of a successful decode as a function of the received symbols similar to that of the standardized Raptor code described above. The decoding failure probability of RaptorQ code can be modeled by (1) [18]:

$$p_{f_{RQ}}(n,k) = \begin{cases} 1, & \text{if } n < k \\ 0.01 \times 0.01^{n-k}, & \text{if } n \geq k. \end{cases} \tag{1}$$

In (1), $p_{f_{RQ}}(n,k)$ denotes the probability of a failed decode of a RaptorQ protected block with k source symbols if n encoding symbols has been received.

4. Online algorithms

Several approaches have emerged for the efficient application of AL-FEC protection utilizing randomized and deterministic online algorithms.

The randomized Algorithm 1 of [12] processes a sequence of packets selecting equiprobably a value from a fair range, which denotes the introduced transmission overhead, when a source block is formed. Subsequently, the introduced transmission overhead is computed according to the random choice of the random variable. In more detail, the randomized online algorithm processes each packet and distributes it in the appropriate AL-FEC source block according to the selected source block length. At the last symbol of each source block the algorithm makes a random choice of the amount of redundant packets the AL-FEC encoder will produce for this particular block. Consequently, the randomized online algorithm applies a random spread of the

Algorithm 1 Randomized AL-FEC Algorithm of [12]

procedure (pkt, sbl)
 $sbn \leftarrow \lfloor pkt.uid/sbl \rfloor$
 if $pkt.uid \bmod sbl \neq 0$ **then**
  $pkt.sbn \leftarrow sbn$
 else
  $pkt.sbn \leftarrow sbn$
  select equiprobably a value i from the set $\{0.05 : 0.01 : 0.5\}$
  transmission overhead $\leftarrow \lceil sbl * i \rceil$
 end if
end procedure

introduced overhead at all of the blocks that the transmitted object is divided into. The competitive ratio for the algorithm of [12] is:

$$c = 1.275 \cdot (1 - p).$$

In [13] the deterministic online Algorithm 2 is presented. The proposed algorithm is based on weights assignment in each processed AL-FEC packet. The algorithm takes as input each processed packet and assigns a weight to the packet according to its unique id i.e., the number of packets included in each FEC source block and the size of the source block each packet belongs to. Thereafter, the algorithm determines if the processed packet will be included in the introduced redundancy comparing the assigned packet's weight with a selected threshold. The value of the threshold determines the required robustness of the AL-FEC protection. Finally, the algorithm examines if the processed packet is the last packet of the current FEC source block in order to compute the transmission overhead will introduce to the multicast transmission. The deterministic online algorithm introduces adaptation features in AL-FEC application in the sense of the transmission overhead reduction as the length of the AL-FEC source block increases. Actually, the assigned weight of each packet reduces with the source block size increase resulting the algorithm to introduce fewer repair AL-FEC symbols as the source block grows for a given threshold.

The competitive ratio for this deterministic online algorithm of is:

$$c = (1 + sbl^{t-1})(1 - p).$$

Algorithm 2 Weighted AL-FEC Online Algorithm of [13]

procedure (pkt, sbl, t)
 $pkt.w \leftarrow log_2(pkt.uid)/log_2(sbl)$
 if $pkt.w \leq t$ **then**
  $count \leftarrow count + 1$
 end if
 if $pkt.id \bmod sbl = 0$ **then**
  transmission overhead $\leftarrow count/sbl$
 end if
end procedure

Finally, in [19] is presented the deterministic online Algorithm 3 that extends the online scheme of [13] and comes to enhance its performance, introducing an adaptive variation based on the outcome of previous multicast deliveries of the transmitted object. In more detail, the proposed adaptive algorithm takes as input a sequence of symbols, assuming one symbol per packet, the length of the source blocks that will be produced and a quantity that represents a threshold. The value of this quantity determines the User Equipments (UEs) coverage that the algorithm should achieve. Furthermore, in each AL-FEC symbol is assigned a weight, with the value of this weight, in conjunction with the value of the threshold, determining if the processed symbol will be included in the computation of the introduced AL-FEC transmission overhead.

The competitive ratio for this deterministic online algorithm is equal to the competitive ratio achieved by the online algorithm of [13] but seems to be more efficient in practice due to its adaptation nature.

Algorithm 3 Adaptive Weighted AL-FEC Algorithm of [19]

```
procedure (symbols, sbl, targetThreshold)
    compute factor
    if factor ≤ targetThreshold then
        threshold ← threshold − (0.05 ∗ threshold)
    else
        threshold ← threshold + (0.05 ∗ (1 − threshold))
    end if
    count ← 0
    for all symbols do
        symbol.w ← log2(symbol.uid)/log2(sbl)
        if symbol.w ≤ threshold then
            count ← count + 1
        end if
        if symbol.uid mod sbl = 0 then
            transmission overhead ← count/sbl
        end if
    end for
end procedure
```

5. Performance evaluation

In this section we present at first the network model and the assumptions we utilize for the evaluation conducted in this work. Thereafter we provide simulation results for the performance of the compared error protection approaches. For the evaluation testbed of this work, we compare the protection performance achieved between two basic error control scenarios. The first scenario assumes that the unicast flows are protected entirely by a retransmission based scheme where a mobile user is able to indicate which data should have been received but have not and request retransmission of missing data. The retransmission of lost data is provided through a point-to-point channel. For the second evaluated scenario we assume that the exclusive error protection scheme utilized for the reliable provision of RTP/UDP flows is a FEC scheme based on RaptorQ codes. For this case we evaluate the application of FEC through the three novel deterministic and randomized online algorithms that were previously described. The main concern of the provided evaluation is the impacts of the amount of packets exchanged between the unicast source and mobile clients, for the successful reception of the transmitted content. We provide simulation results for the performance of those two error protection schemes over several network scenarios.

5.1. Network model

The transmission environment we introduce refers to a typical streaming environment to mobile users. A bunch of data are transmitted to a fraction of mobile users through unicast unreliable radio channel. The transmitted data considered to be a continuous object, as in a streaming delivery session, are encapsulated in RTP/UDP flows, where a source injects packets into the network.

On the AL-FEC protection mechanism, we consider the application of the newly introduced RaptorQ FEC scheme [16]. The sender introduces redundant information within the source data in order to enable receivers to overcome independent packet losses and successfully reconstruct the transmitted data. On the AL-FEC encoding, the transmitted object is partitioned in one or several source blocks. Each FEC source block consists of k source symbols with k depending on the selection of the encoding parameters. The size of a FEC source block is denoted as source block length (sbl). Through the RaptorQ encoding, for each FEC source block, a certain amount of redundant symbols, also called repair symbols, are generated according to the desired amount of protection introduced by the multicast source. A unique ID is assigned on each resulting encoding symbol, which can be a source or a repair symbol, in order to identify the type of the symbol according to the assigned value. At the receiver side, a multicast client is able to determine, for each FEC source block, which source symbols should have been received but have not and is also able to determine the number of encoding symbols it has received.

In this work, we assume the transmission of a packet sequence with independent packet loss masks applied to each mobile receiver according to an examined packet loss rate. In each packet sequence, each packet is denoted by the triplet $\{uid, sbn, r_{il}\}$ where:

- uid: is a unique ID identifying each AL-FEC resulting packet
- sbn: is the number of the FEC source block the examined packet is organized to
- r_{il}: defines if the examined packet was not received by the receiver i with the boolean l set to 0 if packet was not received.

The behavior of the network is modeled as a loss transcript, consisting of the values of the boolean variables r_{il}. In more detail, in the general mobile network model we consider, the values r_{il} may be set arbitrarily, allowing for bursty periods of loss which need not to be correlated across the receivers. More precisely, the packet loss pattern applied to the sequence of transmitted packets is denoted by p, which is the average network packet loss rate taking values in the range [0, 1]. At each receiver, a packet loss mask is applied independently based on the value of p. Furthermore, we have to denote that the packet erasures are randomly distributed at each receiver.

At each receiver the AL-FEC decoding process is modeled according to the decoding failure probability of (1) in order to denote the examined AL-FEC source block as successfully reconstructed or not. On the decoding process, we assume that a sufficient threshold for the failure probability of a recovered source block is 10^{-2} or less as proposed in [20].

5.2. Simulation results

In this section case we evaluate the application of FEC through the three novel deterministic and randomized online algorithms that were previously described. The main concern of the presented evaluation is the impacts of the amount of packets exchanged between the unicast source and mobile clients, for the successful reception of the transmitted content. We provide simulation results for the performance of those two error protection schemes over several network scenarios.

5.2.1. Number of packets

In the first part of the provided simulation results we illustrate the total amount of data exchanged in the mobile network for different values of simulated packet loss rate. In more detail, in Fig. 1 we present the total number of packets exchanged in a mobile network of 100 UEs that receive an object of 1024 packets over unreliable unicast bearers evaluating the average packet loss rate in the range of 1%–20%. For the feedback-based error recovery case, we assume that each UE requests the retransmission of the lost packets until all the required packets have been successfully received. For the evaluation of the FEC-based error control cases,

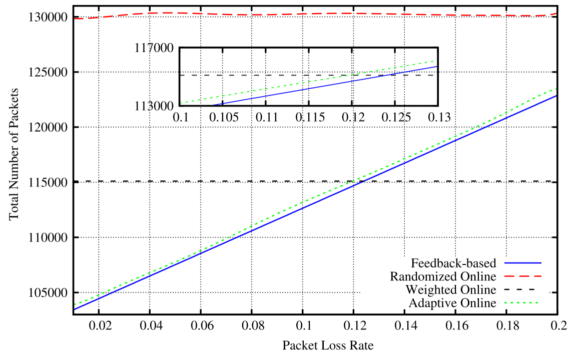

Fig. 1. Total number of packets vs. packet loss rate.

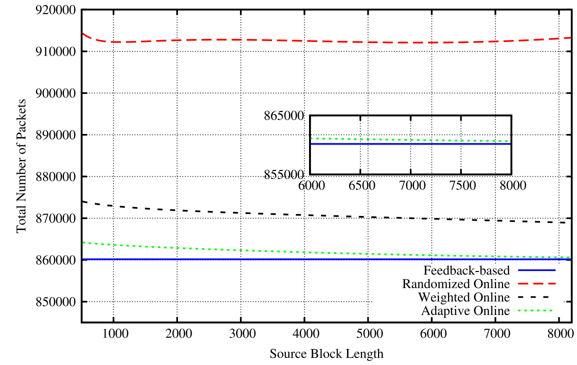

Fig. 2. Total number of packets vs. source block length.

we assume that the transmitted object is partitioned in 4 source blocks each one of length 256 symbols. The results for the randomized online AL-FEC algorithm refer to the average number of packets after 10 consecutive simulations for each evaluated value of packet loss rate. The setup for the weighted online AL-FEC algorithm assumes that the selected value of the threshold t is 0.7 while for the adaptive weighted online AL-FEC algorithm the selected value of *targetThreshold* is again 0.7 and the provided results refer to the simulation of 10 consecutive transmission rounds in order the algorithm to reach a converged state.

Regarding the case of the retransmission-based error recovery, we can observe that the total number of packets exchanged in the network increases in proportion to the packet loss rate increase. Each UE participating in the reception of the transmitted object is able to determine which packets should have receive but has not and requests the retransmission of the lost packets through a unicast channel. Obviously, as the average packet loss rate of the network increases, the number of retransmitted packets and as a consequence the total amount of transmitted data increases too. Furthermore, as long as the network packet loss rate increases the number of established retransmission session for each particular UE increases too. Analyzing the curves of the utilization of the evaluated online algorithms for the application of RaptorQ FEC as the primary error protection method, regarding the randomized online algorithm we can immediately remark that the algorithm just introduces random amount of overhead in the transmission. Obviously the randomized algorithm is a naive scheme which simply selects the introduced overhead in a fair range of values and in average it introduces an almost constant amount of overhead close to 25%. This is why we observe that the difference on the amount of transmitted packets, compared to the retransmission-based case, constantly reduces as the packet loss rate increases. For the case of the deterministic weighted online algorithm, we observe that the algorithm introduces a constant amount of transmission overhead for all of the evaluated values of packet loss rate. This is something anticipated since, the online algorithm adapts the introduced transmission based on the size of the length of the AL-FEC source blocks the transmitted object is partitioned to and since it is a feedback-free scheme cannot make any adaptation on the packet loss rate conditions. On the other hand, the last online algorithm, the adaptive weighted algorithm, which is an extension of the previously described online scheme, we observe that is able to adapt the AL-FEC transmission overhead to the packet loss rate. Based on this, we can remark that the adaptive online scheme can operate very close to the retransmission-based scheme in the context of transmitted packets with respect to the requested value for the percentage of the "recovered" UEs.

5.2.2. Source block length

In this subsection we provide simulation results for the performance of the evaluated online AL-FEC schemes over different values of AL-FEC source block length. In Fig. 2 we present the total number of exchanged packets trend against the length of the AL-FEC source block. For this evaluation we simulate the transmission of an object of 8192 packets to 100 mobile UEs over unicast bearers. The average packet loss rate is fixed at 5% and the evaluated values of the source block length are {512, 1024, 2048, 4096 and 8192}. Again we assume that the threshold t for the weighted online AL-FEC algorithm is 0.7. The same lies for the setup of the adaptive weighted online AL-FEC algorithm simulating 10 consecutive transmission rounds.

Regarding the behavior of the retransmission-based case, the constant number of transmitted packets is anticipated for this error recovery method since there is no AL-FEC encoding applied on the transmitted object and therefore the data are not partitioned in source blocks. For the case of the randomized online algorithm the increase of the source block length cannot have any impacts on the introduced amount of AL-FEC redundancy since the algorithm just applies a random selection on the introduced transmission overhead. The interesting part of the presented results refers to the performance achieved by the two weighted deterministic online algorithms. We can immediately observe that the weighted online algorithm achieves improved performance in terms of the amount of data transmitted as the source block length increases. This behavior directly implies from the operation concept of the weighted online algorithm as well as from the performance properties of the RaptorQ FEC code. Finally, we can remark that the adaptive online algorithm combines its adaptation nature with the weight assignment process based on the source block length and is able to reach the performance of the retransmission-based method as the size of the source block increases.

5.2.3. Satisfied UEs

In this paragraph we provide simulation results evaluating the amount of satisfied UEs against the length of the AL-FEC source blocks. The evaluated values of the source block length are {512, 1024, 2048, 4096, 8192, 16 384, 32 678 and 65 536}. A satisfied UE is defined as a UE that is able to reconstruct the received stream according to the amount of source data that was received. Fig. 3 presents how to amount of satisfied UEs, in terms of percentage, varies with respect to the AL-FEC source block increase. We simulate the transmission of a RTP/UDP stream to 100 mobile UEs over unicast bearers with the average packet loss rate of the access network fixed at 5%. Regarding the online algorithms setup the threshold t of both the weighted online AL-FEC algorithm and the adaptive weighted online AL-FEC algorithm is 0.7 and we simulate 10 consecutive transmission rounds for the case of the adaptive weighted algorithm.

The simulation results presented Fig. 3 reveal the impacts of the application of the two examined error protection schemes on the amount of satisfied UEs receiving the transmitted data.

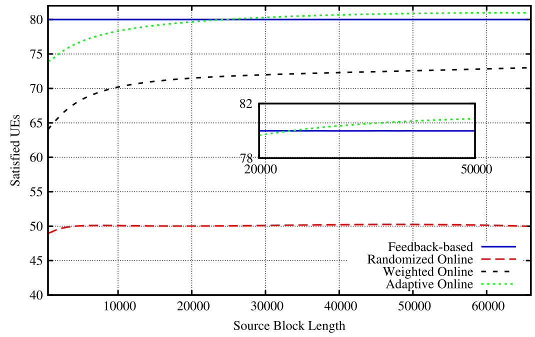

Fig. 3. Percentage of satisfied UEs vs. source block length.

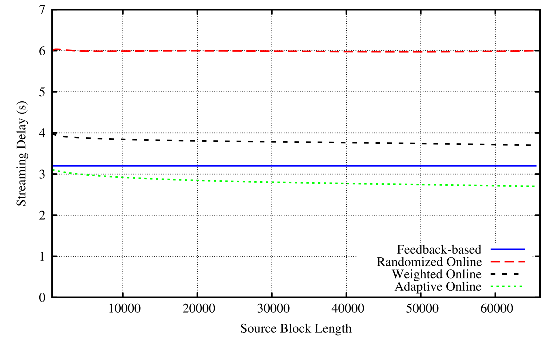

Fig. 4. Average streaming delay vs. source block length.

Regarding the case of the AL-FEC protection scheme based on the online algorithms we can extract some very interested remarks from the presented plots. At first, we can observe that the randomized online algorithm presents the worst performance of all the evaluated cases and its performance is not affected by the source block length increase. This is something anticipated since the randomized online algorithm just randomly selects a value from a predefined range in order to select the introduced AL-FEC redundancy. Therefore, the introduced redundancy remains in average fixed and is not affected by the source block length value. On the case of the weighted deterministic algorithm we observe that its achieved performance on the amount of satisfied UEs is enforced by the AL-FEC source block length increase, and it can achieve almost 10% higher performance when the source block length is increased from the lowest evaluated value to the highest. This is a direct impact of the algorithm's operation since it is able to adapt the introduced AL-FEC redundancy according to the selected source block length. However, its performance is away from the performance achieved by the adaptive weighted algorithm and the feedback-based error correction scheme. The deterministic adaptive online algorithm exploits its ability to monitor the outcome of the previous transmission rounds in order to adapt the introduced redundancy to the network packet losses conditions and therefore is double enforced exploiting also the inherited ability from the weighted algorithm to adapt the AL-FEC transmission overhead according to the selected source block length. We observe that the performance of the adaptive algorithm is very close to the performance of the feedback-based scheme and after the value of 32 768 for each AL-FEC source block it even exceeds the performance of the feedback-based scheme on the satisfied UEs. Finally, regarding the feedback-based scheme it is expected that its performance will not be affected by the source block length since this is an AL-FEC encoding parameter which is never applied in the case of the feedback-based approach.

5.2.4. Playback delay

In this last part of the provided simulation results we evaluate the performance of the two examined error protection approaches regarding the impacts on the playback delay of the simulated RTP flow. The provided results refers to the average value of the playback delay noticed in all UEs participating in the delivery of the streaming content. In the case of streams that are protected from the AL-FEC scheme we should also take into account the tune-in delay for the streaming playback delay. Tune-in delay is defined as the time interval between the start of the packets reception until the start of correct decoding the received packets of each FEC source block. Tune-in delay is experienced by a user who joins the streaming session and the first received packet is anywhere but at the very start of the FEC source block. On the tune-in process a receiver first synchronizes to the FEC block, waiting for the reception and successful processing of each FEC block, before attempting to decode the media. Subsequently, the tune-in delay is a function of the FEC protection period and the decoding delay, typically defined as *tune-in delay* $=$ *protection period* $+ \varepsilon$ [21]. It is obvious that tune-in delay strongly depends on the FEC encoding parameters and more specifically on the selected length of the FEC source block and the introduced AL-FEC transmission overhead. The presented results refers to the simulation of the transmission of a RTP flow of H.264 stream with 128 kbps video source rate considering the non-interleaved packetization mode. For this evaluation we simulate the transmission of a RTP/UDP stream to 100 mobile UEs and the threshold t of both the weighted online AL-FEC algorithm and the adaptive weighted online AL-FEC algorithm is 0.7 simulating 10 consecutive transmission rounds for the case of the adaptive weighted algorithm.

In Fig. 4 we provide simulation results for the average streaming delay of the examined UEs against the increase of the AL-FEC source block length from the size of 512 until the value of 65 536 symbols and the average packet loss rate is fixed at 5%.

Regarding the results provided for an individual time constraint property of the streaming delivery i.e., the average playback delay of the transmitted stream on all of the UEs participating on the delivery we can observe some very interesting results for the performance of the online AL-FEC application approaches and especially for the application of the adaptive weighted online algorithm. At first we have to notice once again that the randomized online scheme does not provide any improvements on the achieved delay since as it has already noted in the previous simulation results the increase of the selected source block length for the AL-FEC encoding has no impacts on the selection of the introduced AL-FEC redundancy of the algorithm due to its operation nature. Hence it is normal the randomized algorithm to present one again the worst performance of all the examined schemes. Regarding the weighted online algorithm we can observe that the algorithm is able to reduce the delay as the source block length increase since, and this is a direct consequence of the algorithm's logic which reduces the introduced AL-FEC transmission overhead as the source block length increase. Therefore, since less AL-FEC packets are transmitted with the source block length increase it is anticipated that the playback delay of the stream will be reduced too. Regarding the two last schemes that present the best performance among the evaluated schemes, once again we do not expect any differences on the performance of the feedback-based approach with the AL-FEC source block size as already noted before and we present in the plot the performance of the feedback-based scheme for comparison purposes. A very interesting remark is that the adaptive weighted online algorithm achieves better performance from the feedback-based approach which is enhanced with the AL-FEC source block length increase due to its operation concept as it has already analyzed in the previous paragraphs. The better performance of the adaptive online scheme in contrast to the feedback-based

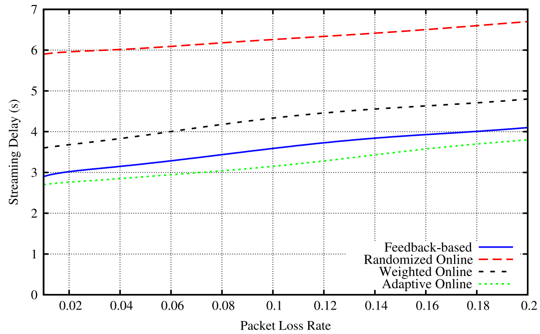

Fig. 5. Average streaming delay vs. packet loss rate.

scheme is a direct consequence of the AL-FEC protection properties since AL-FEC is a feedback-free error protection technique which benefits the individual time constraints of streaming delivery in contrast to the feedback-based approaches which add a time consuming overhead for the process of lost packets retransmission requests and their actual transmission.

At this last part, in Fig. 5 we present simulation results for the average streaming delay with respect to the average packet loss rate of the network with the examined values varying between 2% and 20% and the source block length fixed at 32 768 for the AL-FEC encoding.

Finally, we examine the impacts of the average packet loss rate increase in the average playback delay of the simulated stream. We observe that as the packet loss of the network increase the streaming delay increases too for all of the examined approaches of this work. This fact is due to the increasing number of lost packets on each individual receiver participating on the streaming delivery and is not relevant with the operation concept of each error-protection scheme. Once again the plots confirm the performance properties that we have analyzed in previous simulation results for each examined scheme. Furthermore, we can observe that the adaptive weighted online algorithm is able to achieve better performance regarding the time constraints of a streaming delivery compared to the feedback-based approach under different reception conditions of the network.

6. Conclusions

In this work we have examined the opportunity of utilizing AL-FEC protection as the primary and the only error protection method over mobile unicast streaming services. Since the reduction of the AL-FEC application over mobile multicast networks to an online problem was newly introduced we have grasped the opportunity to examine the impacts of the application of online algorithms for the application of AL-FEC protection against a common feedback-based error control method. We have examined the performance of three different online algorithms, randomized and deterministic, aiming on the efficient application of AL-FEC application against the performance of the common method of error control, i.e., a retransmission-based scheme.

At first we have presented the evaluated online algorithms and we have analyzed their operational concepts. Thereafter, we have introduced the network model under which we have conducted the presented evaluation, which refers to a typical mobile network where data are transmitted to multiple mobile users through unreliable unicast bearers. Thereafter, we have provided and analyzed simulation results for the performance achieved by the evaluated error control schemes in terms of the total amount of data transmitted in the network.

Regarding the outcome of the conducted simulations, the most interested results came up from the performance achieved by the adaptive weighted online algorithm. This deterministic scheme is able to adapt the introduced AL-FEC transmission overhead based on the length of the source block the transmitted object is partitioned too as well as the reception conditions of the network. This fact implies that the adaptive weighted online algorithm is able to exploit the performance properties of the utilized RaptorQ FEC code and at the same time to adapt the AL-FEC overhead according to the individual packet losses of each recipient. Based on the simulation results, we were able to verify that this online scheme is able to operate close enough to the performance of a retransmission-based error recovery method. Furthermore, we have to remark that with a careful selection of the AL-FEC encoding properties the online scheme can achieve almost the same performance with the feedback-based method.

7. Future work

Some possible future steps that can follow this work are a more comprehensive evaluation of the online schemes for the AL-FEC application considering also other network parameters and settings. Furthermore, the design of more sophisticated and dedicated on unicast environments online algorithms for the AL-FEC policy online problem could be beneficial for the efficient application of AL-FEC protection over mobile unicast services. It is our belief that the newly introduced approach which utilizes online algorithms for the design of an efficient application policy for the AL-FEC protection, provides a strong basis for the design development of algorithms that can address the main impairments of the AL-FEC protection under different delivery scenarios.

References

[1] 3GPP Multimedia Broadcast/Multicast Service (MBMS); Protocols and codecs (Release 10), TS 26.346, 3rd Generation Partnership Project (3GPP), 2011. URL http://www.3gpp.org/ftp/Specs/html-info/26346.htm.

[2] ETSI, Digital Video Broadcasting (DVB); Guidelines for the implementation of DVB-IPTV Phase 1 specifications; Part 3: Error Recovery; Sub-part 2: Application Layer - Forward Error Correction (AL-FEC), TS 102 542-3-2, ETSI, 2011.

[3] M. Watson, T. Stockhammer, M. Luby, Raptor Forward Error Correction (FEC) Schemes for FECFRAME RFC 6681, Aug. 2012. URL http://www.ietf.org/rfc/rfc6681.txt.

[4] R. Motwani, P. Raghavan, Algorithms and Theory of Computation Handbook, Chapman & Hall/CRC, 2010, (Chapter). Randomized algorithms. URL http://dl.acm.org/citation.cfm?id=1882757.1882769.

[5] A. Borodin, R. El-Yaniv, Online Computation and Competitive Analysis, Cambridge University Press, New York, NY, USA, 1998.

[6] Y. Bartal, J. Byers, M. Luby, D. Raz, Feedback-free multicast prefix protocols, in: Computers and Communications, 1998. ISCC'98. Proceedings. Third IEEE Symposium on, 1998, pp. 135–141. http://dx.doi.org/10.1109/ISCC.1998.702473.

[7] J. Janssen, D. Krizanc, L. Narayanan, S. Shende, Distributed online frequency assignment in cellular networks, J. Algorithms 36 (2) (2000) 119–151. http://dx.doi.org/10.1006/jagm.1999.1068, URL http://www.sciencedirect.com/science/article/pii/S0196677499910684.

[8] L. Lin, N. Shroff, R. Srikant, Asymptotically optimal energy-aware routing for multihop wireless networks with renewable energy sources, IEEE/ACM Trans. Netw. 15 (5) (2007) 1021–1034. http://dx.doi.org/10.1109/TNET.2007.896173.

[9] W. Liang, X. Quo, Online multicasting for network capacity maximization in energy-constrained ad hoc networks, IEEE Trans. Mob. Comput. 5 (9) (2006) 1215–1227. http://dx.doi.org/10.1109/TMC.2006.133.

[10] A. El Gamal, C. Nair, B. Prabhakar, E. Uysal-Biyikoglu, S. Zahedi, Energy-efficient scheduling of packet transmissions over wireless networks, in: INFOCOM 2002. Twenty-First Annual Joint Conference of the IEEE Computer and Communications Societies. Proceedings, Vol. 3, IEEE, 2002, pp. 1773–1782. http://dx.doi.org/10.1109/INFCOM.2002.1019431.

[11] I. Caragiannis, A.V. Fishkin, C. Kaklamanis, E. Papaioannou, Randomized on-line algorithms and lower bounds for computing large independent sets in disk graphs, Discrete Appl. Math. 155 (2) (2007) 119–136. http://dx.doi.org/10.1016/j.dam.2006.04.036.

[12] C. Bouras, N. Kanakis, V. Kokkinos, A. Papazois, Deploying AL-FEC with Online Algorithms, in: Next Generation Mobile Apps, Services and Technologies, NGMAST, 2013 Seventh International Conference on, 2013, pp. 175–180. http://dx.doi.org/10.1109/NGMAST.2013.39.

[13] C. Bouras, N. Kanakis, A competitive AL-FEC framework over mobile multicast delivery, in: Wireless Communications and Mobile Computing Conference, IWCMC, 2013 9th International, 2013, pp. 305–310. http://dx.doi.org/10.1109/IWCMC.2013.6583577.
[14] C. Bouras, N. Kanakis, Online AL-FEC protection over mobile unicast services, in: European Conference on Networks and Communications, EuCNC, IEEE, Paris, France, 2015, http://dx.doi.org/10.1109/EuCNC.2015.7194074, June 29-July 2, pp. 229–233.
[15] H. Schulzrinne, A. Rao, R. Lanphier, Real Time Streaming Protocol (RTSP), RFC 2326 (Informational), Apr. 1998. URL http://www.ietf.org/rfc/rfc2326.txt.
[16] M. Luby, A. Shokrollahi, M. Watson, T. Stockhammer, L. Minder, RaptorQ Forward Error Correction Scheme for Object Delivery, RFC 6330, Aug. 2011. URL http://tools.ietf.org/rfc/rfc6330.txt.
[17] M. Luby, A. Shokrollahi, M. Watson, T. Stockhammer, Raptor Forward Error Correction Scheme for Object Delivery, RFC 5053 (Proposed Standard), Oct. 2007. URL http://www.ietf.org/rfc/rfc5053.txt.
[18] 3GPP, Rationale for MBMS AL-FEC Enhancements, Tdoc S4-110449, 3rd Generation Partnership Project (3GPP), 2011.
[19] C. Bouras, N. Kanakis, An adaptive weighted online AL-FEC algorithm over mobile multicast networks, in: Wireless Communications and Networking Conference (WCNC), 2014 IEEE, IEEE, 2014.
[20] 3GPP, Simulation results for the performance and complexity of RS codes for MBMS FEC, Tdoc S4-050107, 3rd Generation Partnership Project (3GPP), 2005.
[21] 3GPP, Report of FEC selection for MBMS, Tdoc S4-050250, 3rd Generation Partnership Project (3GPP), 2005.

Christos Bouras is Professor in the University of Patras, Department of Computer Engineering and Informatics. Also he is a scientific advisor of Research Unit 6 in Computer Technology Institute and Press—Diophantus, Patras, Greece. His research interests include Analysis of Performance of Networking and Computer Systems, Computer Networks and Protocols, Mobile and Wireless Communications, Telematics and New Services, QoS and Pricing for Networks and Services, e-learning, Networked Virtual Environments and WWW Issues. He has extended professional experience in Design and Analysis of Networks, Protocols, Telematics and New Services. He has published more than 450 papers in various well-known refereed books, conferences and journals. He is a co-author of 9 books in Greek and editor of 2 in English. He has been member of editorial board for international journals and PC member and referee in various international journals and conferences. He has participated in R&D projects.

Nikolaos Kanakis was born in Kalamata, Greece in 1985. He obtained his Diploma from the Computer Engineering and Informatics Department of Patras University, Greece in 2009. In 2011 he received his Master Degree in "Computer Science and Technology" and in 2014 he received his Ph.D. on Reliable Multicasting over Mobile Cellular Networks from the same department. He is working in the Research Unit 6 of Computer Technology Institute and Press "Diophantus" participating in the *AutoBAHN* development team. His research interests include next generation mobile telecommunications networks, multicast protocols and reliable multicasting. He has published several research papers in well-known referred international journals and conferences.