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# A$^3$L-FEC-FSFB: Age-Aware Application Layer Forward Error Correction with Fixed Sampling Rate and Fixed Block-length

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*Abstract*—**The concept of Age of Information (AoI) has emerged as a measure of data freshness in status-update-oriented applications. Enhancing AoI often involves unconventional networking strategies that adjust update transmission rates based on network conditions and resource variability. However, most prior research on AoI has been theoretical, assuming ideal network models and known delay/service time distributions. In real-world networks, obtaining these statistics and optimizing for them is challenging. In this study, we introduce Age-Aware Application Layer Forward Error Correction (A$^3$L-FEC), a data flow controller at the application layer of the OSI model. A$^3$L-FEC aims to enhance data freshness by preventing age violations, achieving this through reliable transmission of fresh data using forward error correction and the user datagram protocol.**

*Keywords—Age of Information (AoI), Age violation, Communication protocol, Age-aware congestion control, A$^3$L-FEC.*

## I. INTRODUCTION

In recent times, there has been a notable increase in the demand for transmitting the real-time status of dynamic systems through reliable communication channels. This demand spans various sectors such as industrial automation, network management, and remote monitoring, where accurate and timely tracking of system status holds significant importance. Maintaining an acceptable system response time is crucial to ensure proper and timely monitoring of the system's status. However, in situations of network overload where the volume of data surpasses the network's capacity, the response time of the network or its queues tends to increase, resulting in diminished performance and quality. Elevated response times typically manifest as increased queuing delays and packet losses, leading to congestion within the network. Essentially, congestion presents a formidable obstacle to the efficient and timely transmission of variable system status packets across networks. Thus, it becomes essential to devise strategies for congestion management in networks, particularly in scenarios requiring fresh data transmission for system monitoring.

This study introduces A$^3$L-FEC-FSFB: Age-Aware Application Layer Forward Error Correction with Fixed Sampling Rate and Fixed Block-length, a pragmatic communication network congestion control algorithm. A$^3$L-FEC leverages forward error correction (FEC) and the user datagram protocol (UDP). This algorithm regulates age violations along the end-to-end path by dynamically adjusting the rate of status updates injected into the network. The underlying concept is to ensure an adequate flow of status updates during transit to provide fresh data at the destination, while simultaneously preventing a buildup of excess update packets at network bottlenecks. This approach underscores the importance of congestion control algorithms adopting a strategy akin to "keeping the pipe just full, but no fuller," as suggested by Kleinrock, [1].

Various congestion control algorithms have been developed to enhance network performance. The first group, exemplified by Reno and Cubic, relies on packet loss to signal congestion, leading to high throughput but increased queuing delays. A second category, including Vegas and FAST, focuses on using delay as the congestion indicator to optimize network link capacity. However, these algorithms often overestimate delay due to factors like ACK compression, resulting in suboptimal link utilization. To address this, a third category of congestion control algorithms, like Bottleneck Bandwidth and Round-trip propagation time (BBR), aims for accurate delay estimation while maximizing link capacity utilization.

None of the existing congestion control categories prioritize freshness. Fortunately, recent researches have delved into various aspects of AoI to ensure fresh updates reach their destination. Notable studies [2]–[4] have explored AoI in real-world networks and its optimization, including works such as [5]–[13]. For instance, [5] conducted the first emulation study of AoI in wireless links using CORE and EMANE, evaluating AoI under different system settings. [6] measured AoI over TCP/IP links, observing non-monotonic AoI trends. [7] examined AoI in a two-way connection and observed behaviors consistent with theoretical queuing analysis [14]. Furthermore, [8] established Internet and IoT testbeds to study AoI behaviors across wired and wireless links.

Efforts to integrate AoI optimizers into communication systems have been promising. For instance, [9] introduced a deep reinforcement learning-based algorithm for transmission rate optimization, addressing AoI as a Markov Decision Process. [10] proposed WiFresh, a scheduling solution enhancing information freshness in WiFi networks, achieving significant improvements under high loads. [11] developed the Age Control Protocol (ACP) for multi-hop IP networks, aiming to minimize AoI through adaptive sending rates, later refining it with ACP+ [12]. Evaluating ACP and ACP+ on IoT devices has been presented in [13], where it addressed implementation issues, suggesting modifications for specific network types.

***Contribution:*** Our study introduces a protocol aimed at improving AoI within network environments. We prioritize the efficient transfer of dynamic process samples to ensure optimal data freshness. While our approach shares similarities with ACP, it diverges significantly in methodology and execution.

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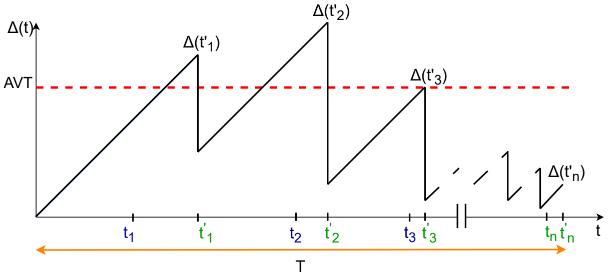

Fig. 1: Sample path of the age process $\Delta(t)$, [15].

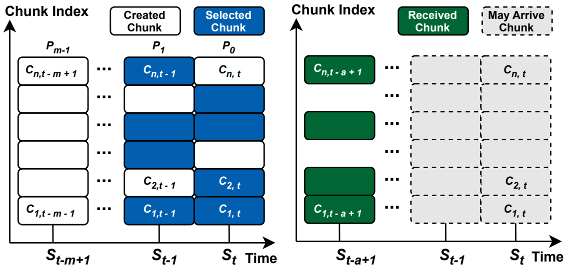

Fig. 3: Chunks in the A$^3$L-FEC-FSFB protocol: Created, Selected, Received, and May Arrive.

## II. SYSTEM MODEL

The status age, denoted as $\Delta(t) = t - t'$, represents the time difference between the current time $t$ and the generation time $t'$ of the freshest received data. It measures the elapsed time since the most recent data was produced. In Fig. 1 of [9], a sample variation of age $\Delta(t)$ is depicted, showcasing a sawtooth pattern over time $t$, starting from an initial age $\Delta_0$ when observation begins at $t = 0$. Each source status update generated at times $t_1$, $t_2$, ..., $t_n$ is received at corresponding times $t'_1$, $t'_2$, ..., $t'_n$. In the absence of updates, the age increases linearly over time but decreases after each update reception. Age violation occurs when the observed age $\Delta(t)$ exceeds a predefined threshold, indicating that the freshness requirement has not been met. In general, the AoI is defined by $\overline{\Delta} = \frac{1}{T}\int_0^T \Delta(t)dt$ and corresponds to the area under the age graph in Fig. 1, normalized by time $T$.

In the context of A$^3$L-FEC-FSFB at the application layer for this study, the system is conceptualized as a time-slotted status update system operating across an error-prone communication link (see Figure 2). A receiver observes a time-varying source process through data received from a remote transmitter over a Packet Erasure Channel (PEC) with restricted link availability. Let $S$ denote the source process, comprising samples $\{s_\tau\}_{\tau\geq 0}$, generated at uniform intervals. Throughout this study, all time durations will be normalized relative to the duration of a single time slot.

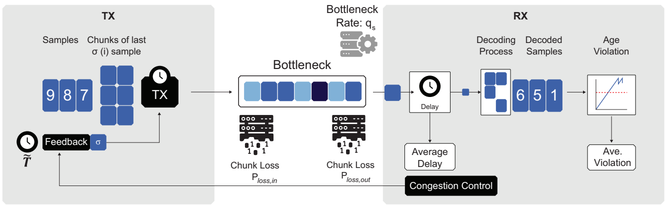

Fig. 2: System model of a status update system over an error-prone link with a transmission rate feedback.

After sampling, the transmitter divides the $K$-bit sample into $k$ equally sized data chunks and encodes these data chunks into $n$ coded chunks of equal length $\frac{K}{k}$ bits using a maximum distance separable (MDS) code. This encoding ensures that the receiver can decode the sample whenever it has at least $k$ coded chunks out of the $n$ available. Denoting the $i$th coded chunk of sample $s_\tau$ generated at time $\tau$ as $c_{i,\tau}$, the transmitter sends these coded chunks to the receiver via the UDP Protocol, allocating one UDP packet per coded chunk. It's assumed that the transmitter can send multiple UDP packets in a single time slot. Let $\mathcal{T}_t$ represent the set of chunks transmitted during time slot $t$, i.e., $c_{i,\tau} \in \mathcal{T}_t$ indicates that the $i^{th}$ coded chunk of sample $s\tau$ is transmitted at time slot $t$. Furthermore, it's assumed that the transmitter only buffers coded chunks from the $m$ most recent samples from the available coded chunks in a given time slot, denoted as $\mathcal{T}_t \subset c_{i,\tau} : i = 1, ..., n :, : \tau = t - m + 1, ..., t$ (Figure 3).

We utilized the M/M/1 queuing model with a first come first serve (FCFS) service discipline. The transmitter forwards UDP packets to the receiver at a rate of $\tilde{\sigma}$ [codeword/time slot] through a single infinite buffer bottleneck, where the queue's service rate is denoted as $q_s$ [chunk/time slot]. Each transmission of a chunk consumes a constant time, equivalent to $\frac{1}{q_s}$ time slots if no chunks precede it in the queue. We assume that UDP packets face a probability $P_{loss,in}$ of being dropped before entering the bottleneck and a probability $P_{loss,out}$ of being lost before reaching the receiver. Consequently, the overall probability that a UDP packet is lost in the network, denoted as $P_{loss,c} = P_{loss,in} + (1 - P_{loss,in})P_{loss,out}$.

Because of the network bottleneck, there's a variable delay before a packet reaches the receiver. Let $D_{t;i,\tau}$ represent the delay of coded chunk $c_{i,\tau}$ between its transmission and successful reception when it's sent during time slot $t$. A UDP packet might be lost or corrupted, in which case the delay is considered infinite, i.e., $D_{t;i,\tau} = \infty$.

At the receiver, successfully received coded chunks are retained and utilized for the recovery of the corresponding codeword (refer to Figure 3). Denote $\mathcal{C}_t$ as the set of coded chunks successfully received by time $t$, and $\mathcal{S}_t$ as the set of samples that can be decoded at time $t$. A sample cannot be decoded at time $t$ only if there are more than $n-k$ missing coded chunks for that sample, $\{s_\tau \notin \mathcal{S}_t\} = \{\sum_{i=1}^{n} \mathbb{1}_{\{c_{i,\tau}\notin\mathcal{C}_t\}} > n - k\}$, where $\mathbb{1}$ is the indicator of a missing coded chunk event, $\mathbb{1}_{\{c_{i,\tau}\notin\mathcal{C}_t\}} = \prod_{j=\tau}^{t}\left(\mathbb{1}_{\{c_{i,\tau}\notin\mathcal{T}_j\}} + \mathbb{1}_{\{c_{i,\tau}\in\mathcal{T}_j\}}\mathbb{1}_{\{D_{t;i,\tau}>t-j\}}\right)$.

In this study, AoI is $\Delta(t) = t - \max\{\tau \in \{1, 2, ..., t\} : s_\tau \in \mathcal{S}_t\}$. Note that for a specified age violation threshold value, $AVT$ [time slot], the goal of the A$^3$L-FEC protocol is to minimize the age violation metric, $AV$. Here, $AV = \frac{1}{T}\sum_{t=1}^{T} \mathbb{1}_{\{\Delta(t)\geq AVT\}}$ is as a measure for the quality of monitoring the process $S$, with $T$ as the monitoring duration.

To enhance monitoring quality, we introduced the *Stationary Independent Selection* (SIS) policy. This policies is crafted to optimize sample selection for fresh data transmission.

**Definition 1.** *A transmission policy is said to be a stationary independent selection policy if the events* $\{c_{i,\tau} \in \mathcal{T}_t\}$ *are independent for all* $i$, $\tau$ *and* $t$ *such that* $\{c_{i,\tau} \in \mathcal{T}_t\}$ *occurs with probability* $p_{t-\tau}$ *where* $p_v = 0$ *for* $v < 0$ *and* $v > m - 1$.

According to this definition, sample selection for transmission follows the principles of stationarity and independence. This implies that the occurrence of events, denoted as $c_{i,\tau} \in \mathcal{T}_t$, is regarded as independent for all $i$, $\tau$, and $t$. Furthermore, a sequence of probabilities $p_0, ..., p_{m-1}$ characterizes a SIS policy in line with this definition.

In our threshold-based AoI framework, transmitting outdated status updates is avoided by actively filtering out samples exceeding $AVT$, ensuring only fresh data is sent. This approach enhances information freshness, and reliability in the monitoring process. While the transmission rate is constant ($\tilde{\sigma}$) and the relationship is as follows: $\sum_{j=0}^{m-1} p_j = min(\tilde{\sigma}, AVT)$. Let $\Pi_{SIS}$ represent the set of all possible SIS policies. Now we will focus on addressing the following problem:

$$\min_{\pi \in \Pi_{SIS}} \limsup_{t \mapsto \infty} \frac{1}{T} E\left[\sum_{t=1}^{T} I_{\Delta(t) > AVT}\right] \tag{1}$$

In particular, for a given expected transmission rate $\tilde{\sigma} = min(\tilde{\sigma}, AVT)$, the A$^3$L-FEC-FSFB protocol chooses the $\mathbf{p}^* = [p_0^*(\tilde{\sigma}) ... \, p_{m-1}^*(\tilde{\sigma})]^T$ such that:

$$p_j^*(\tilde{\sigma}) = \begin{cases} 1 & \text{if } j \leq \lfloor \tilde{\sigma} \rfloor, \\ \tilde{\sigma} - \lfloor \tilde{\sigma} \rfloor & \text{if } j = \lfloor \tilde{\sigma} \rfloor + 1, \\ 0 & \text{if } j > \lfloor \tilde{\sigma} \rfloor + 1. \end{cases} \tag{2}$$

At any time slot $t$, the value $p_j^*(\tilde{\sigma})$ represents the probability of selecting any encoded chunks of the sample generated at time slot $t - j$ for transmission at time slot $t$. Here, $\tilde{\sigma}$ denotes the number of codewords sent per time slot, where each codeword comprises $n$ chunks.

It is important to emphasize the transmission rate is calculated at the receiver and sent to the transmitter every $\tilde{T}$ time slots as feedback. This feedback is used by the age violation (3) and average chunk delays (4). Where, $AV_{MI}$ signifies the average number of age violations within the $MI^{th}$ monitoring interval, computed by summing $\mathbb{1}_{\Delta(t) > AVT}$ for each time slot $t$ within the interval, then dividing by $\tilde{T}$. Meanwhile, $\bar{W}_{MI}$ represents the average chunk delay experienced by transmitted chunks during the $MI^{th}$ monitoring interval and calculated as the ratio of the $L_1$ norm of the delay set, $\mathcal{D}_{MI}$, to its cardinality. Note that $MI \in \{1, 2, 3, ..., \lceil \frac{T}{\tilde{T}} \rceil\}$; and $\mathcal{D}_{MI}$ denotes the set of delays of all chunks of all codewords that are successfully received by the end of $MI^{th}$ monitoring interval.

$$AV_{MI} = \frac{1}{\tilde{T}} \sum_{t=(MI-1) \times \tilde{T}}^{MI \times \tilde{T}} \mathbb{1}_{\{\Delta(t) > AVT\}}. \tag{3}$$

$$\bar{W}_{MI} = \frac{\|\mathcal{D}_{MI}\|}{|\mathcal{D}_{MI}|} \tag{4}$$

## III. CONGESTION CONTROL OF A$^3$L-FEC-FSFB

In communication systems, every transmitted packet enhances data freshness, prompting the receiver to desire an increased transmission rate. However, excessive packet injection can lead to congestion, impacting delay and age. To address these issues, the transmitter should adjust its transmission rate to fill the network optimally. The congestion control of A$^3$L-FEC-FSFB, depicted in Fig. 4, incorporates an adaptive rate mechanism comprising various blocks, each tasked with specific functions. Some blocks are activated under urgent conditions, while those within the green and red zones are triggered during regular conditions.

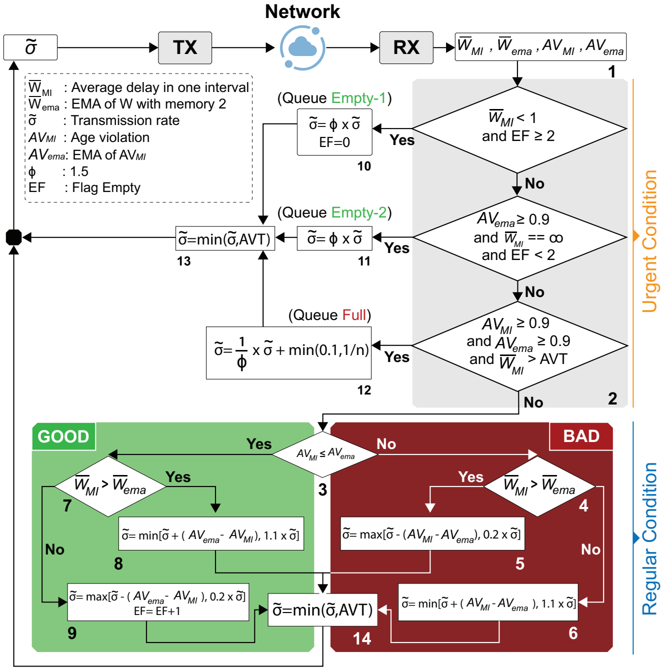

Fig. 4: Update algorithm of A$^3$L-FEC-FSFB protocol.

## IV. A$^3$L-FEC AND THEORETICAL LIMITS

**The upper bound for transmission rate ($\sigma^{up}$):** It denotes the maximum transmission rate that can be supported without causing instability in the bottleneck of the network. It acts as a reference point to determine the optimal transmission rate, where $q_s$ denotes the service rate of the bottleneck, indicating the maximum rate at which packets can be processed at the bottleneck of the network, $P_{loss,in}$ represents the packet loss probability at the bottleneck input, $n$ represents the number of encoded chunks into which a sample is divided. Indeed each sample is encoded into $n$ number of chunks.

$$\sigma^{up} = q_s \times \frac{1}{n \times (1 - P_{loss,in})}. \tag{5}$$

**The lower bound for age violation:** To establish a lower bound for age violation, we assume the presence of an ideal queue, (i.e., $q_s = 0$) and evaluate the average age violation for a given transmission rate $\sigma$. Under this assumption, the sample $s_\tau$ which is generated at time slot $\tau$ is decoded with probability $P_{d_{s_\tau}}(z)$ until time slot $z$ as follows:

$$P_{d_{s_\tau}}(z) = \sum_{i=0}^{n-k+1} \frac{n!}{(n-i)!i!} (P_l(z-\tau))^i (1 - P_l(z-\tau))^{n-i} \tag{6}$$

$$P_l(z - \tau) = (P_{loss,c})^{z-\tau+1}, z \geq \tau \tag{7}$$

**Outage probability:** Let's denote the event $E_{e,t}$ as a scenario where the age of the system at time $t$ is $e$ [time slot]. The probability of observing the $E_{e,t}$ is expressed as:

$$P(E_{e,t}) = P_{d_{s_e}}(t) \times \prod_{j=t-e+1}^{t} (1 - P_{d_{s_j}}(t)) \tag{8}$$

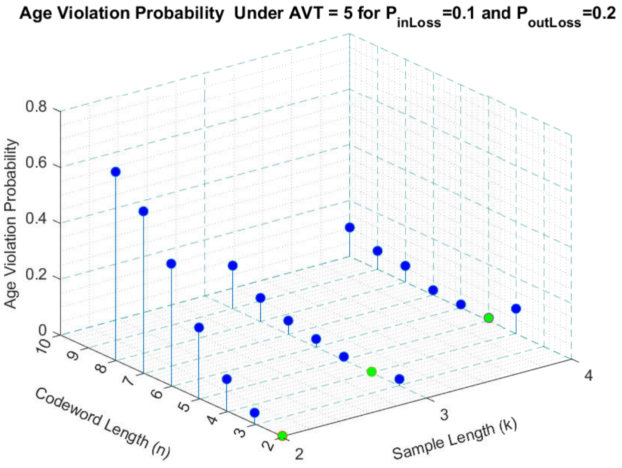

Fig. 5: Simulation results under different coding rates.

Utilizing (8), we derive the probability of the age exceeding "$e$", (9). This probability is named the outage probability.

$$P_{Outage}(Age > e) = 1 - \sum_{i=0}^{e} P(E_{i,t}) \qquad (9)$$

## V. CODING RATE AND A$^3$L-FEC-FSFB

To assess how the coding rate affects the performance, we ran simulations based on a system model akin to that in Section II. Transmissions used an FCFS queue with service rates of $k \times 1.4706$ packets per time slot and a queue size of 5000 chunks, with a propagation delay of 1 time slot. Simulations lasted $10^5$ time slots, with monitor intervals of $10^2$ time slots. We tested two age violation thresholds: 2 and 5, under varying packet loss probabilities. Each scenario was repeated 50 times and averaged. This approach provided a reliable understanding of the system's performance. The results indicate that selecting an appropriate coding rate, represented by the green point, notably improves the system's age violation performance. This improvement is particularly pronounced in scenarios with high packet loss probabilities. Due to space constraints in the paper, we only presented simulation results for age violation with a threshold of 5 across different coding rates, specifically when $P_{inLoss} = 0.1$ and $P_{outLoss} = 0.2$ in Fig. 5.

## VI. A$^3$L-FEC VS. ACP+: PERFORMANCE COMPARISON

We compared the performance of A$^3$L-FEC-FSFB with ACP+, [12], through simulation results. In most simulations, A$^3$L-FEC-FSFB consistently outperforms ACP+. Results show that our proposed algorithm consistently maintains transmission rates lower than the bottleneck service rate capacity (i.e., the optimal transmission rate). However, these selected transmission rates remain close to the optimal transmission rate calculated using (5). This approach ensures the algorithm avoids significant queuing in the buffer while providing fresh data to the client. For the ACP+ protocol, we observed that the chosen transmission rate rarely exceeds the queue capacity. However, there are instances where the ACP+ operates at very low rates, resulting in high age violation values.

## VII. CONCLUSIONS

This study introduces a novel data flow control algorithm named Age-Aware Application Layer Forward Error Correction with Fixed Sampling Rate and Fixed Block-length (A$^3$L-FEC-FSFB), specifically designed to prioritize freshness in data communication systems. The protocol aims to enhance data freshness, measured by the age-violation metric, in real-time communication scenarios over TCP/IP networks. Through comparisons with existing algorithms like ACP+, the effectiveness of A$^3$L-FEC in addressing freshness challenges in various scenarios is demonstrated. As future work, we plan to introduce the A$^3$L-FEC with Variable Sampling Rate and Variable Block-length (A$^3$L-FEC-VSVB) and assess its performance against different TCP congestion control flavours.

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