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FEC Methods' Effects on Intelligent Video Streaming Over 2.4 GHz Wireless Channels: An Analysis

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Abstract: The increased usage of portable and cellular devices in recent decades has made streaming videos over wireless networks more challenging. The 2.4 GHz wireless channels' non-linear nature causes erratic video transmission due to interference from other wireless devices and general channel noise. Jitter can result from network issues and uncorrected data loss at the decoder, leading to packet loss. MPEG-compressed videos require significant bandwidth, posing another challenge. To address this, a model was introduced to measure MPEG-4 video streaming with forward error correction (FEC) efficiency in wireless local area networks (WLANs) using IEEE 802.11 Distributed Coordination Function (DCF). The model considers the pattern of packet loss in the wireless network, including diffused packet loss and packet loss. Simulation results show that the suggested approach has higher throughput and reduced packet loss compared to the current technique.

Keywords: *Packet loss, channel congestion, fuzzy controller, and MPEG encoding.*

1. Introduction

The use of videos in everyday life as a primary source of entertainment is now prevalent. To maximize the storage space when transferring and storing video, video compression is crucial. Wireless video transmission is currently regarded as one our daily life's most engaging applications. In communications network, users are granted access to the final mile of connectivity through wireless networks. Despite the flexibility and user mobility offered by these networks, video transmission over them faces significant problems [1]. There are still huge concerns with enhancing multimedia applications quality of service (QoS) of, in addition to other issues that Wireless networks face such interference, noise, and bandwidth variations.

Due to the fact that the majority of conventional service structures, network topologies, and protocols were not created with high data rates and real-time digital video transmission demands in mind, some of them are unable to provide a reliable distribution route [2]. Both 2.4 GHz and 5 GHz are often used in Wi-Fi networks, and one has advantages over the other.

Among the 2.4 GHz's key benefits is that it is not only less expensive to produce devices using this frequency , but also the 2.4 GHz waves can easily pass through solid surfaces like walls and floors. Consequently they gain more popularity [3]. In a wireless network, fading is one of the key issues [4]. When the received signal is found to consist of many attenuated, delayed, and phase-shifted copies of broadcast signals, multipath propagation is the likely culprit. In the vector sum of these several components, each of which coming via a separate path, is the final signal that is received. Due to the random nature of how these components accumulation, the received signal's amplitude can alter in either a positive or negative way depending on their relative phase differences.

Wireless networks constrained and constantly shifting bandwidth is another issue. Wireless networks can handle high data speeds, although they often have a small scale. Another challenge with wireless channels is interference. The effectiveness and scalability of wireless

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connections are typically decreased as a result. The bandwidth of wireless channels is constrained. Even though such networks are not ideal for streaming video directly over them, uncompressed video has a very high bandwidth. For more robust video communication through noisy channels, several techniques have been put up to address these concerns. These include hierarchical modulation, adaptive modulation, joint source channel coding (JSCC), automatic repeat request (ARQ) retransmission, adaptive source channel coding, robust source coding, and forward error correction (FEC) [5]. The quantity of video sent across at least one wireless hop is likely to keep growing as cellular internet rates continue to rise and wireless channel is being used by more users. To provide flawless video playing for receivers, this kind of application requires efficient routing methods with high bandwidth, and content delivery techniques. The existing wireless networks will probably use a variety of access technologies and Internet technology to function. In wireless contexts, achieving effective bandwidth aggregation faces a number of difficulties relating to usage, network instability, network congestion, and connection variability and power usage. This research uses a sophisticated transmission strategy for adjusting the controller to the wireless channel's current status and maintain MPEG video quality throughout the communication flow. In the architecture, two fuzzy controllers - the fuzzy controller with rules and a neural-fuzzy controller - monitor the input-output buffer to maintain traffic shaping and provide the MPEG encoder with appropriate parameters for sending video across the network.

In order to link devices to the Internet, wireless networks have emerged as one of the most crucial methods, boosting efficiency and promoting information sharing. Wi-Fi, also known as IEEE 802.11, has emerged as the dominant standard for wireless local area networks. Latency, jitter, and packet loss are the three most crucial parameters for assessing Wi-Fi quality. Packet loss can happen for a variety of reasons and occurs when a few packets fail to arrive at their destination. Application quality as perceived by users over Wi-Fi networks is impacted by packet loss, particularly for real-time and multimedia apps. A development for the more effective techniques to the network design and performance analysis, in addition more accurate simulated using computers, is made possible by the introduction of explicit Wi-Fi network models for packet loss. Given that packets might be lost due to a variety of factors, such as buffering problems, noise, multipath, signal attenuation, signals distortion, thermals noise, and conflict to the access media, modelling packet loss at these networks is a significant task. In this paper, a thorough discussion of the models that may be used to simulate packet lossing in the Wi-Fi networks is presented as well as an explanation of why packets get lost. The survey's possible benefits are as follows: (i) a thorough overview of the parameters of the various scenarios of packet drop in Wi-Fi networks, together with thorough examination of the packet drop rate for each model, (ii) a comparison of the models in light of input parameters and verification scenarios, and (iii) a summary of unanswered questions and potential research topics. The key elements Wi-Fi networks' packet loss process, as well as the advantages and disadvantages of the most popular models for packet losses, will hopefully be better understood by researchers with the aid of our analysis.

A collection of rules for the WLANs is called IEEE-802.11. (WLANs). Since its introduction in 1997, the IEEE 802.11 standard has undergone constant revisions to enhance, among other things, security, dependability, throughput, and Quality of Service. [6]. The IEEE 802.11a/b/g/n specifications for WLANs are part of Wi-Fi (wireless fidelity), which permits consumers to access the internet at high speed. [7], not to mention the most recent variations, such IEEE 802.11af/ac (2013) and IEEE 802.11ax (2019) [8]. A proliferation of Wi-Fi capable devices as a result of improvements in wireless communication networks has significantly increased, which has in turn encouraged the creation of new, less expensive instruments and software with lower power requirements [9]. Worldwide mobile device users now have access to a wider range of services and applications thanks to quick advancements in wireless data transmission. Wi-Fi networks are widely used and can be found in a variety of locations, including hotels, airports, public parks, and retail establishments. Wi-Fi primarily supports file sharing, web surfing, streaming audio and video, chatting, and email [6]. Emerging video coding methods, such as 8K resolution and scalable video coding, are predicted to cause a considerable growth in data traffic for video streaming [10]. In 2022, mobile networks and Wi-Fi will carry 71% from IP traffic, as estimated by Cisco [11]. Wi-Fi network carried 43% from each IP traffic in the world in 2017. The model is a simulation or the condensed depiction for the real or hypothetical systems which intended to the highlight key elements of the system under study, prediction, modification, or control [12]. As a result, while not all of the modelled system is represented in the model. A model, according to Fournier [13], is a perfect illustration meant to correctly reflect all pertinent characteristics of the initial mechanism and typically contains stochastic elements. When a packet loss model offers the system under study, practical perceptions, forecasts, and solutions, it is beneficial [12]. Since the 1960s, there have been packet drop scenarios for digital communications that have been put forth in an effort to simulate the behavior of packet loss in actual networks. Insightful frameworks that can represent as the process of packet drop have been found to be beneficial in studying the functionality of wireless networks, and modelling and simulation approaches are vital for knowing how wireless systems perform [14]. Early models for packet loss solely took physical layer faults into account. Noise, multipath leakage, Low signals power, and interference, are the primary physical-layer causes from the losing packet at Wi-Fi networks [15-16]. However, there are a variety of physical and link-layer issues that can lead to packet drop at the Wi-Fi networks. In Wi-Fi network, buffer overflows, buffer bloats, queuing delays, conflicts as well as harmful attacks are the main sources of packet loss at the link layer. It may be challenging to determine whether packet dropping in the latter scenario are indeed the consequence of malicious assaults or other factors [17]. Significant performance gains may result from reducing the effects of packet loss, particularly for real-time applications like voice conferencing [18–20] and live video streaming [21]. The paper is organized so that the introduction comes in the 1st section, followed by the approach and execution steps in the 2nd section, and the results in 3rd section. The fourth section contains the conclusion as well as a few suggestions.

2. Methods and Implementation

Information about the study approach and its application is provided in this section. For clarification and consistency, various words are also defined. Regardless to the state of the network's links, judgements are made to enable input data, choose which packets to reject, and schedule tasks at different systemic locations. On the other hand, decisions about feedback are founded on the idea of evaluation. This strategy is broken down into three elements for managing congestion [22]. By keeping an eye on your system, you can spot the time and place where congestion happens, reroute information that can be put to use, and alter the performance of the system to solve problems.

2.1 Intelligent Video Transmission

A neuro-fuzzy diagram in Fig. 1 depicts the overview for video transmissions. Each video frame is subjected to spatial compression. Out of the overall video’s quality, which affected by the level of spatial compressions, the models here place the emphasis on video bitrate adjustment. Image quality and compression are typically trade-offs. Nevertheless, the quality and compression rate of the video, as well as the bitrate rate, is a key consideration for selecting an acceptable spatial compression strategy [23]. Quantization and the Discrete Cosine Transform (DCT) are crucial aspects of spatial models. Using the DCT method, N x N blocks of data are compressed into a spatial frequency weighted sum. The following equation expresses the 2D DCT [24-25] of an N×N block.

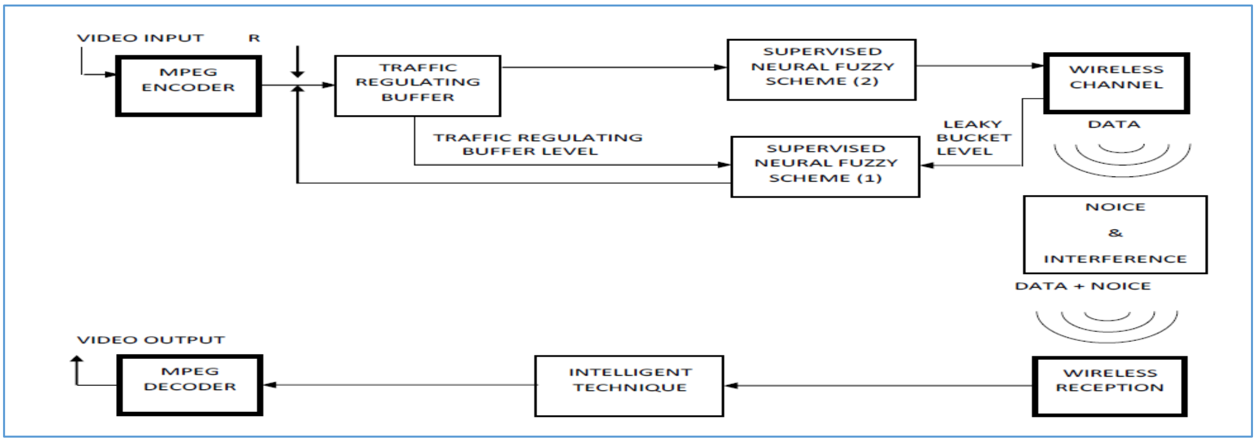

Fig. 1: Schematic diagram for video streaming with neural fuzzy technique.

2.2 Packet Loss Modeling

Since the 1960s, researchers have investigated the issue of network packets loss modelling. Primary packet losing models are discussed at this part, with focusing on models that apply to the Wi-Fi networks. 1st proposed packet drop model (such as the Gilbert-Elliot and Gilbert models) were created for wired networks before being adapted to wireless ones. The most recent models are specifically made to consider Wi-Fi network behavior in the instance of abrupt data loss. With additional wireless networks like ZigBee and LTE, many of the models that have been presented can also be deployed.

2.3 Analysis Model

This section details the analysis model used to study effect for packet loss in the transmitted videos quality.

2.3.1 MPEG Group of Pictures (GOP)

P-frames, I-frames, and B-frames are the three frame types specified by the MPEG [26] standard for compressed video streams. MPEG I Frames (Intra Frame Coding) are individually encoded and decoded. In a video series, prediction of the preceding I or P frames is used for encode MPEG Predictive Coding (P) frames. Measuring the preceding I or P frames is used to encode MPEG B (Bidirectional Predictive Coding) frames. Generally, the term "Group of Images" refers to the standard method of dividing a video sequence into reduced blocks that can subsequently be encoded together (GOP). GOP mode is defined by the two parameters G and (N, M). N and M stand for the distances between I and IP frames, respectively. As illustration , G(9, 3) in Figure 2 shows that the GOP is made up of 1 I-frame, 2 P-frames, and 6 B-frames. The beginning of the subsequent IDP is indicated by the second I-frame, as shown in Figure 2. Arrows depict that I or P frames are required for decoding B and P frames, whether they came before or after them.

2.3.2 Decodable Threshold Network

Video frames broken into smaller packet depending on a maximum size of a network packet prior to their transmission over a decodable threshold network. When at least a specific portion of the packet is received at the video frame, the frame deemed as well decodable. The Decodable Threshold (DT) is the term referring to this estimation [27]. The decoder, for instance, cannot withstand packet loss if DT = 1.0. That is, it just takes one missing packet for a video image to become unusable. Yet, when DT = 0.75, a video frame is indeed considered as viable to be decoded even though 25% of its packets are likely to be dropped from the network. When transmitting the video, video reconstruction techniques like FEC can be employed if the decoder is tolerant of some loss (DT 1.0). Achieving loss tolerance involves adding more data to the video stream (FEC redundancy). For convenience of use, video recovery concerns are not taken into account in this study and DT is set to 1.0. Consequently, this research depicts a worst-case situation related to video transmission quality. Video pictures can be regarded as being either directly or indirectly undecodable based on the layered MPEG coding structure in Figure 2. Lack of video frame packets received inhibit the forward frame from being decoded, as seen by the error message. Alternatively, indirect indecipherability of a video image arises when an image is deemed indecipherable because another image on which it relies fails to be directly decoded.

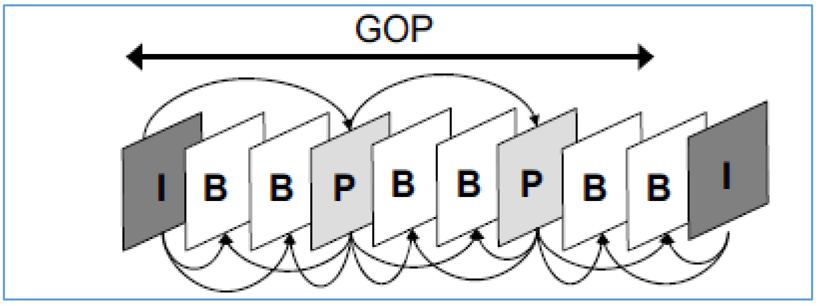

Fig. 2: MPEG GOP (N=9, M=3) example.

2.3.3 Decodable Frame Rate (DFR)

An application layer parameter called Decodable Frame Rate (DFR) used for gauge a quality of the video stream. Range of DFR values from 0 to 1.0. The end user experiences greater video quality with higher value of DFR. The DFR or the proportion of frames that can be decoded provided by video encoder to the overall frame count, is defined as DFR in this context.

$$DFR = \frac{NF_{dec}}{(NF_{total-I}+NF_{total-P}+NF_{total-B})} \quad 1$$

Where NF_{dec} is a summation of NF_{dec-I}, NF_{dec-P}, and NF_{dec-B}. Table 1 illustrate the DFR sign. ddepends on the MPEG coding GOP structure of Figure 1, the formulation and calculation of the decodable frame rate are presented in [27].

Table 1: Adopted Notation.

$NF_{total-I}, NF_{total-P}, NF_{total-B}$	Total number of frames of each type.
$NF_{dec-I}, NF_{dec-P}, NF_{dec-B}$	Number of frames to decode in each type.
NF_{dec}	Total number of frames decoded from the video stream.
NF_{GOP}	Total number of GOPs in the video stream.

2.4 Wireless Channel Error Model

Gilbert [28] suggested in 1960 that a first-order Markov chain be used to simulate continuous bit loss in a burst noise channel. One popular channel model used to assess clustering error patterns is the Gilbert model. In order to create the Gilbert-Elliot (GE) model, Elliot [29] expanded the Gilbert model in 1963 to take into account the loss probability of two states. The state diagram for the GE channel model is displayed in Figure 3. Gilbert takes into account the unique scenario of a completely good state (k = 1), which is represented by the model's two states, "good" (G) or "bad" (B). Reporting is also frequently referred to as "income" and "loss" [30]. The probability of falling in state B is 1 - h, where 0 ≤ h ≤ 1 and the transition from state G is flawless. The variables p and q denote the probabilities of transitioning between states G and B and between states B and G, respectively. To the Gilbert model by k = 1, the Packet Loss Ratio (PLR) defined via the parameter

$$PLR = \left(\frac{p}{p+q}\right)(1-h) \quad 2$$

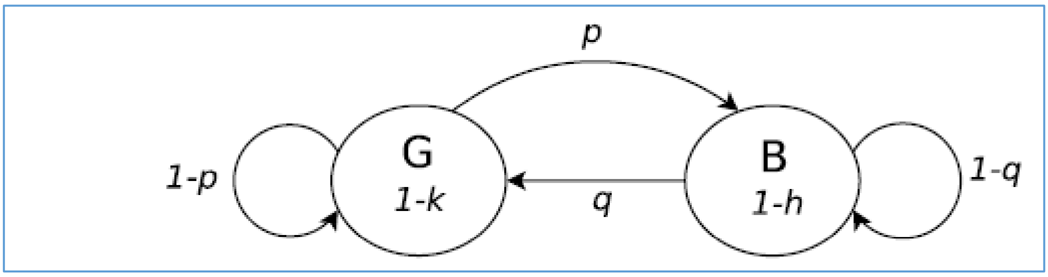

Fig. 3: Channel model of Gilbert-Elliott

Where p is the likelihood that state G will change to state B, and q is the likelihood of the reverse transition. Losses are independent occurrences that can happen in every state with probability. 1 - k and 1 - h, $0 \leq h \leq 1$ and $0 \leq k \leq 1$ for states G and B [31]. Any value for k and h may be selected [32]. In general $p + q < 1$, but the model is shrunk to a Bernoulli model when $p + q = 1$. The transition probability matrix P is defined by the equation

$$p = \begin{array}{c c} & \begin{array}{cc} G & B \end{array} \\ \begin{array}{c} G \\ B \end{array} & \begin{pmatrix} 1-p & p \\ q & 1-q \end{pmatrix} \end{array} 3$$

The fixed probability of the states G and B given via the formulas $\pi_G = q / (p + q)$ and $\pi_B = p / (p + q)$, correspondingly. The PLR is obtained using fixed probability [33-34] also, is defined as

$PLR = (1- k)\ \pi G + (1- h)\ \pi B$

3. Video Stream Performance Analysis Considering FEC Error Correction

FEC error correction's method impact on viewed MPEG-4 video stream quality at IEEE 802.11 DCF WLANs in different networks is evaluated using an analysis model, as suggested in Performance Analysis of Video Streams Utilizing FEC Error Correction [35]. Playable frame rate (PFR) versus FEC overhead versus bit error rate (BER) are the evaluation conditions explained in Figure 4. Ratio of FEC duplicate frames to the total number of frames (source + FEC duplicates) is referred to as FEC overhead. As FEC overhead appears to increase, video quality improves since all BER values have larger probabilities of successful decoding. Even if numerous duplicate packets are injected into the transport stream under poor link conditions (BEP = 10^{-4}), the video quality will still be noticeably reduced. This is because the transport stream is overloaded with packets, both original and duplicate.

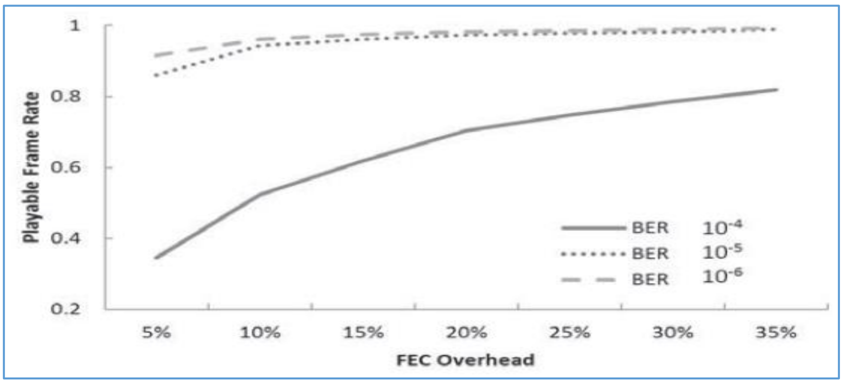

Fig. 4: Change PFR to FEC overhead according to BER [35].

Figure 5 depicts the influence of FEC overhead on a video stream's overall quality when various network parameter loads are present. Video quality is largely independent of duplicate frames number added to the original frame after loading network is regulated (e.g., $n \leq 10$). Yet, because there are more active stations and a higher chance of collisions, the higher FEC overhead leads to lower effective frame loss rates and better video quality.

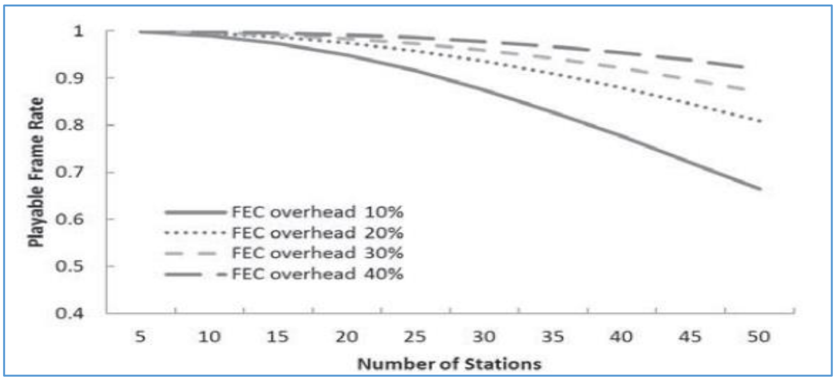

Fig. 5: Change PFR to number of stations according to FEC overhead [35].

4. Simulation Results

The proposed analysis model is examined at this section by comparing results of the analysis produced via the model suggested in [35] with those indicated by the PFR results. A video stream transmitted over a wireless LAN that is FEC-protected is taken into account in the proposed model's PFR performance. The frequency of collisions is proportional to the network's node count. The PFR is reduced as indicated in Figure 6 by more collisions causing more packet loss. As this figure demonstrates The PFR predictions making use of the study's model are more in line with the simulation findings when compared to the outcomes obtained using the analytical model described in [35]. The majority of frame loss in unfavorable channel conditions is due to a radio link failure. Figure 6 demonstrates that all stations can transmit the majority of packets successfully when given more retransmission possibilities (for instance, m = 6). Consequently, as can be shown in Figure 6, the analytical design model beat the model proposed in [35] in terms of PFR prediction.

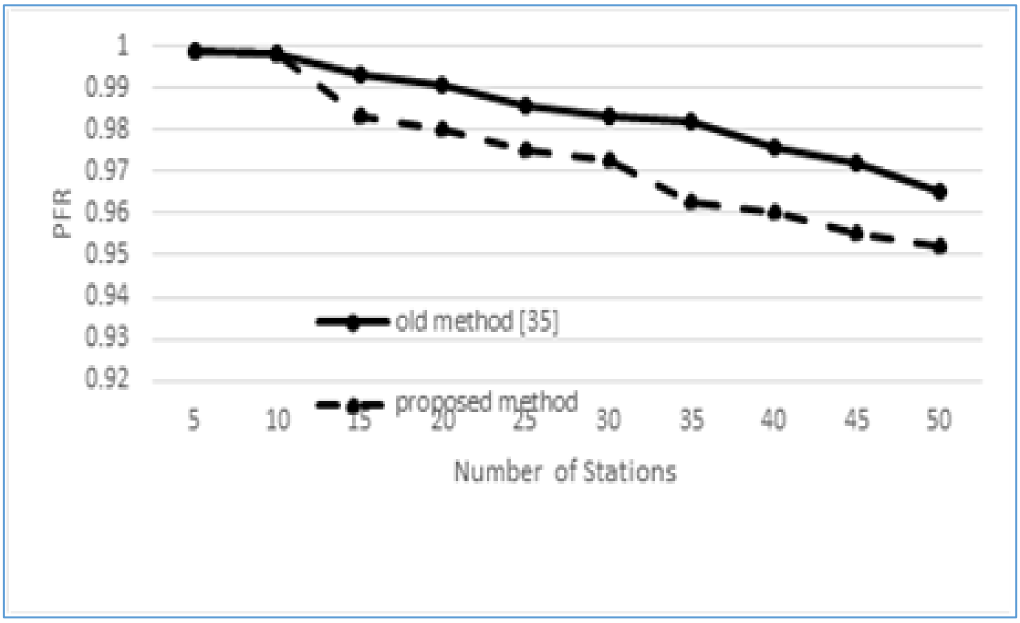

Fig. 6: PFR changes depending on the number of active stations, taking into account the possibility of retransmission of frames.

PFR versus bit error rate (BER) with FEC overhead is shown in Figure 7. When the proportion of FEC superfluous frames to total frames is taken into consideration defines FEC overhead (source plus FEC redundancy). As anticipated, for all BER values, video quality is better when FEC overhead rises because successful decoding is more likely. Unfortunately, because there are so many packets lost during transmission, both primary and superfluous, the video quality will still be significantly reduced even with a huge count of duplicated packets were introduced into the transport stream. Figure 7 demonstrates that the suggested approach has a better FEC recovery mechanism. To demonstrate how losing packet in wireless networks are distributed, the second experiment compared the suggested method with the analytical method presented in [19] under the identical circumstances. Similar to [19], under the identical packet loss scenario, the effect of packet loss on the quality of the provided video is not as significant as that of distributed packet losing. This occurs as distributed packet loss results in a higher rate of frame loss than packet loss. Figure 8 demonstrates that the suggested model's simulated image quality is superior to that of the current model created in [19]. Hence, the analytical approach gives MPEG video transmission over wireless networks known quality restrictions. Another attempt to determine how the amount of the replay buffer affects the caliber of the generated video.

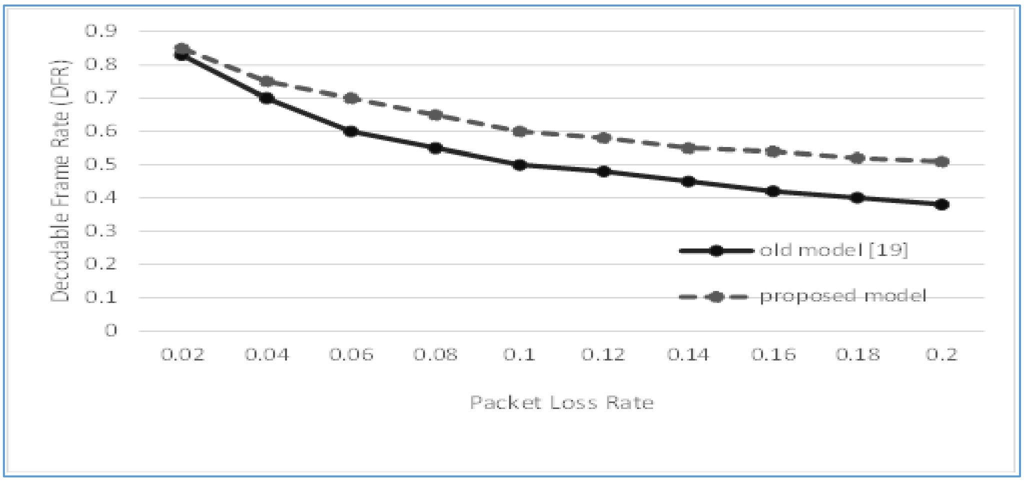

Fig. 7: Change PFR to FEC overhead according to BER.

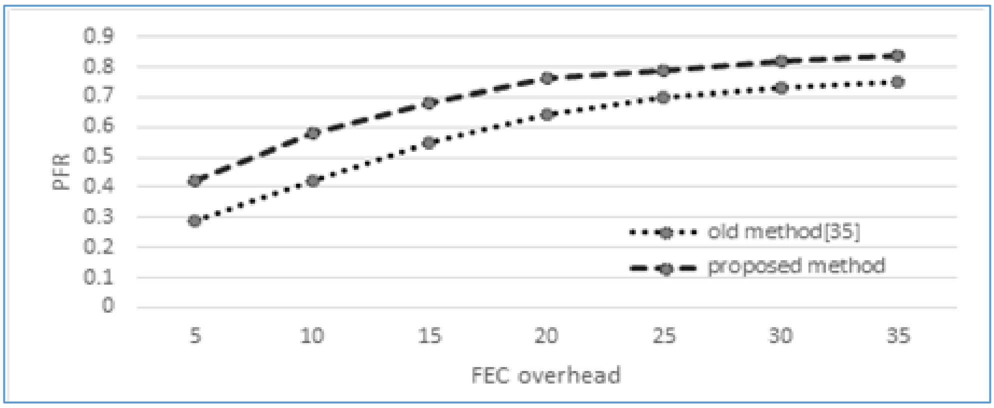

Fig. 8: The DFR of GE error model.

High jitter can result in frame delays that are greater than tolerable end-to-end delays because it impacts the end-to-end video frame arrival time. The receive-side playback buffer in multimedia communication serves to minimize jitter and smooth out user-played video. Figure 9 illustrates how the video quality degrades as the playback size grows. This is because more video frames can be discarded and endpoint latency is reduced when the playback buffer is smaller. The most recent experiment was designed to compare video quality with the model produced in [19] under identical conditions and investigate how transmitted packet size affects video quality. The video transmission quality is influenced by the largest packet size, as illustrated in Figure 10. Video frames are split up into smaller packets for delivery during video transmission constructed on network's the largest packet size. If video transmission does not have forward error correction (FEC) recovery protection, sending small packets can lead to a high rate of frame loss. These simulation results demonstrate that the most important frames may more fully decode subsequent frames if all packets have the same loss probability. The proposed FEC recovery mechanism is ultimately found to be more effective than the approach created in [19].

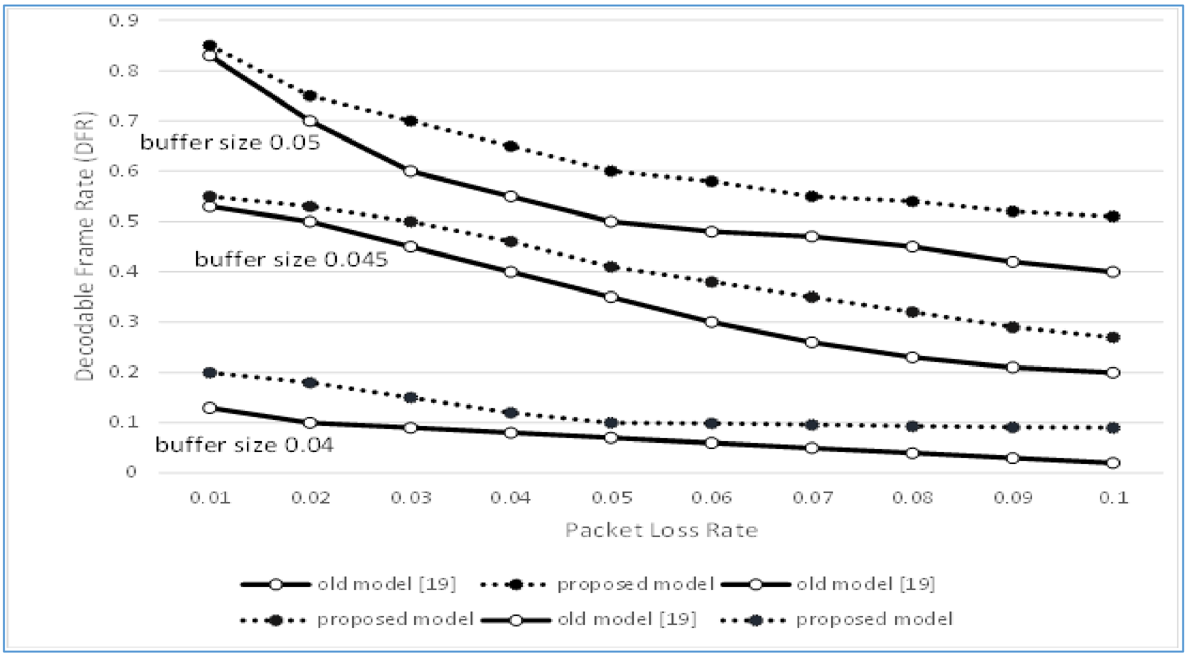

Fig. 9: Poor video quality due to playback buffer.

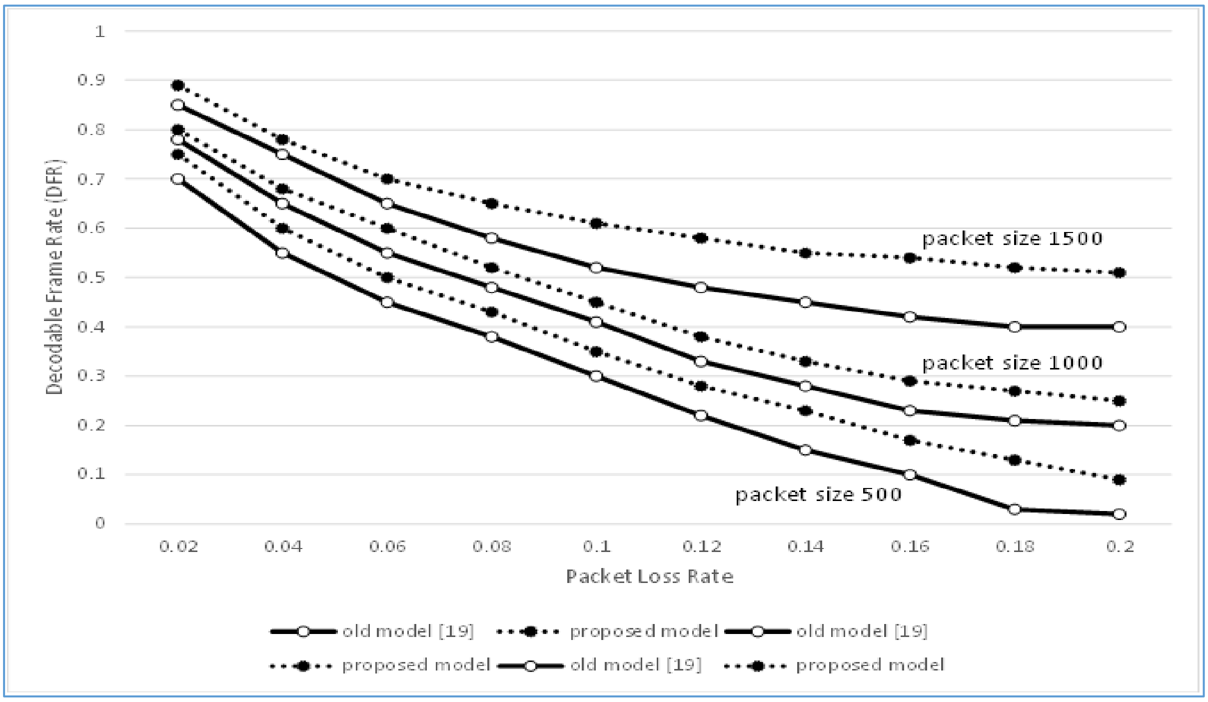

Fig. 10: DFR of different transmission packet size.

5. Conclusions

This study employs a rule-based fuzzy controller and a neuro-fuzzy controller to regulate the arrival and departure rates of traffic shaping buffers in order to enhance image quality by minimizing the standard deviation of MPEG video data loss and improving the overall data spread and Group of Pictures (GOP) size for data transport. The article investigates the impact of packet loss on the reliability of MPEG video transmission over wireless networks, while proposing a new framework for evaluating MPEG-4 streaming video quality using IEEE 802.11 DCF Wireless LAN protocol with forward error correction (FEC) protection. The developed model considers both the performance of the FEC error correcting algorithm in improving the observed video quality at the receiving end, as well as the impact of congestion and frame loss in the wireless network. By comparing the reproducible frame rate (PFR) results obtained from modeling in MATLAB with the two existing analytical models proposed in [19, 35], the proposed model is demonstrated to be effective. The results support the assertion that the suggested model can accurately estimate the perceived quality of MPEG-4 video streams in DCF WLAN with FEC protection. The suggested model will be extended in future research to incorporate interference scenarios in IEEE 802.11e-based network channels that support Quality of Service (QoS).

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