Quality of Video Streaming: Taxonomy

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Abstract: - Real-time and live video streaming are very important topics in networks nowadays, either wired or wireless, and so it is significant to address and study the behavior, advantages, and disadvantages of different techniques and algorithms. This article presents a comprehensive overview for researchers who are willing to conduct research in video compression standards, error correction algorithms for improving the quality of video streaming, forward error correction codes with feedback and forward error correction algorithms with unequal loss (or error) protection techniques for enhancing video streaming quality, description about the layered video streaming, single and multi-paths video streaming, good description about video streaming over wireless networks, the problem of erasure packets and packet erasure networks/packet erasure Channels, layered coding compression techniques, error detecting and correcting algorithms, Unequal Error Protection (UEP) techniques and schemes, multipath video streaming, and recent researches that based on hybrid solutions over 3G, 4G, 5G, WiMAX, and Wi-Fi wireless networks.

Key-Words: - AI, compression techniques, error correction, layered video streaming, video streaming, wireless network.

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1 Introduction

Due to fast life, huge, and successive steps in technology, there is urgent demand to enhance the quality of real-time and live video streaming; whereas live video streaming (such as live YouTube, live news channels, and football matches) can tolerate delay up to ten seconds, the real-time video streaming (such as video conferencing, remote medical consultation, WhatsApp video calling, and remote surgery) cannot tolerate delaying more than 400 milliseconds (ms) between the time of requesting and playing the video streaming, [1], [2].

Many developments and enhancements have been introduced to video and video signal processing that increase the quality of video signals and solve the issues of network overloading and traffic congestion in Replicated-Stream Multicast Video Distribution [2] and [3].

In the beginning, scientific research aimed at solving the problems of packet lost and congestion in video streaming by using Backward Error Correction (BEC) algorithms such as Transmission Control Protocol (TCP), [4]. Where Forward Error Correction (FEC) was used to handle the disadvantages of BEC, [5]. To achieve a high Peak Signal Noise Ratio (PSNR), [6], Unequal Error Protection (UEP) algorithms were used. Hybrid algorithms that merge between advanced BEC and Rateless FEC techniques were adopted to solve the disadvantages of real-time and live video streaming over wireless networks (3G, 4G, 5G, Wi-Fi, and WiMAX), [7], [8], [9].

This work highlights clear and sufficient ideas about the current and most important techniques that are used to enhance the live and real-time video streaming quality over wired and wireless lossy packet networks. Therefore, this work can help students and researchers in their new research.

The remaining of this work is organized as follows: Section 2 compares this work with similar related works and shows both advantages and disadvantages. Then section 3 presents a brief history of video and video streaming, section 4 presents an overview of video streaming compression standards, section 5 presents error correction techniques for enhancing video streaming quality, section 6 demonstrates the effectiveness of FEC codes in enhancing the quality of video streaming, section 7 presents the implementation of UEP techniques in enhancing the quality of video streaming, section 8 discusses the layered video streaming, section 9 describes the difference between single and multiple paths video-streaming algorithms, section 10 presents the benefits of real-time and live video streaming in wireless networks such as 3G, 4G, 5G, Wi-Fi and WiMAX. Section 11 discusses the problem of erasure packets. Finally, section 12 provides the summary, conclusion, and future work.

2 Literature Review of Related Works

In this section, we make a brief comparison between this work and similar related works to highlight both advantages and disadvantages.

[10] reviewed live and real-time video streaming over different old wireless networks. Furthermore, they showed some old video streaming compression techniques. However, they did not present advanced FEC Codes with UEP Techniques. Also, they did not show the recent video compression techniques that depend on Artificial Intelligence (AI). Finally, they did not highlight how to enhance real-time video streaming over recent wireless networks (like: WiMAX, 4G, and 5G). Then, authors in [11] reviewed the enhancement quality of live and real-time video streaming, Furthermore, they highlighted the most important Quality of Experience (QoE) management methods for video streaming services, presenting how various machine learning techniques can be applied to perform proactive control. However, [11] did not show sufficient information about video compression techniques and enhancing the quality of video streaming by using FEC algorithms. Finally, they did not show how to enhance the video streaming quality by using the UEP algorithms.

[12] reviewed and analyzed the development in video streaming such as compression techniques, streaming protocols, cloud video processing, 4K/8K, and showed aspects of future development, that will help to provide quality of service of video streaming and enhance the revenue for service providers. On the other hand, [12] did not highlight real-time video streaming, furthermore, it did not show how to enhance the quality of Video Streaming by using FEC algorithms with Adaptive UEP methods, Finally, [12] did not present the recent video compression techniques that depend on Artificial Intelligence (AI). As a matter of fact, our work will put all the important and recent real-time and live video streaming techniques in one basket which will help the computer and network's researchers in their new research.

After that, [13] reviewed the state-of-the-art in video compression techniques and algorithms. Furthermore, it showed some combination of machine learning and adaptive compression techniques. However, it did not clearly show how to enhance the quality of video streaming using the FEC methods with UEP Techniques. Moreover, it did not present how to enhance the quality of Video Streaming over Wireless Networks (WiMAX, 4G, and 5G).

[14] reviewed live and on-demand video streaming, also they showed the compression and AI techniques that are employed in video streaming applications. On the other hand, they did not clearly show the FEC methods with Adaptive UEP Techniques, Furthermore, they focussed on live video streaming so what about real-time video streaming. Eventually, they did not review the quality of Video Streaming over WiMAX, 4G, and 5G Wireless Networks.

This work will review a full live and real-time video streaming's lifecycle and will put all the important and recent real-time and live video streaming techniques in one basket. Therefore, it can help researchers in their research

3 Short History of Video and Video Streaming

This section provides a brief history of the development of video and video streaming. Videotape Recorder (VTR) was the first video recording which was developed in 1951.In 1976, JVC and Panasonic companies developed the Video Home System (VHS), which was then replaced by– Digital Video (DV) taps and Digital Versatile Disc (DVD). High-speed Digital Signal Processors (DSPs) and Single Instruction stream Multiple Data stream (SIMD) algorithms were used to process video signals [15] and [16].

Color Graphics Adapter (CGA) was introduced in 1981 by IBM company as the first color display standard for Personal Computers [17], in 1987, the display hardware standard - Video Graphics Array (VGA) was developed by IBM, [18]. In 1991, DSP processors were used for video signal processing, [15].

In 1993, a significant development was achieved in Moving Picture Experts Group-2 (MPEG-2) applications, in 1994, MPEG-2 was considered as set-top boxes for digital cable TV and Video on Demand (VOD) [15], VOD was based on packet-switched Asynchronous Transfer Mode (ATM), it was a very vital application in the UK due to the supplement of up to 250 homes as well as schools, [19].

[2] proposed a bandwidth control mechanism that aimed at solving the problems of congestion and network overloading in Replicated-Stream Multicast Video Distribution.

In 1997, researchers developed the first Practical Real-time video streaming that depended on the H.263 compression technique, which allowed users to play videos on the Web [3].

The most important developments and enhancements in video streaming quality started after 1997 that depended on BEC and FEC techniques.

4 Video Compression Standards

Compression is defined as the process of reducing the size of data by detecting and removing redundant data. Compression techniques are classified into two types: Lossless compression and lossy compression., The compressed data in lossless compression can be reconstructed exactly as the original one without any loss, while in lossy compression, the compressed data cannot be reconstructed exactly as the original one. Lossy compression achieves a higher compression ratio than lossless compression, and so lossy compression techniques are more suitable for video streaming applications. This section focuses on different schemes of lossy compression used in video applications. Furthermore, it highlights some video compression techniques that depend on Artificial Intelligence (AI).

4.1 Motion Picture Experts Group-1 (MPEG-1)

In November 1991, MPEG-1 was approved as the first public standard of Motion Pictures. Frames in MPEG-1 are encoded in three different approaches: I-frames (Intra-coded frames) which are encoded as separate frames and independent of adjacent ones. P-frames (Predictive-coded frames) are encoded by prediction from previous I-frame or P-frame; and so, this approach achieves a better compression ratio than the I-frames method. Finally, B-frames (Bi-directional frames) are encoded by prediction from one previous frame and one subsequent frame of either I-frames or P-frames (Figure 1). MPEG-1 is best used in video on digital storage media, [20].

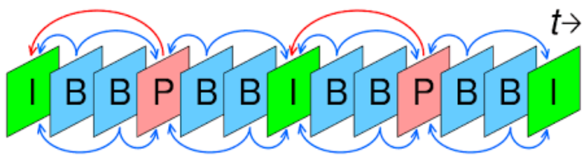

Fig. 1: I Frame, P Frame, and Bi Frame

4.2 Motion Picture Experts Group–2 (MPEG-2)

In 1994, MPEG-2 was approved to enhance the drawbacks of MPEG-1 by handling considerable pictures and achieving higher quality at the cost of using higher bandwidth. It has many applications such as high-definition TV (HDTV), Interactive Storage Media (ISM), and Cable TV (CATV) [21]. Additionally, the compression technique of DVD movies was dependent on MPEG-2. However, MPEG-2 was not fabricated for video applications through the Internet because it required too much bandwidth, [20], [21], [22].

4.3 Motion Picture Experts Group–4 (MPEG-4)

The MPEG-4 project was approved in 1998. One of its important features is achieving a high video compression rate that is used for mobile phone video applications and video applications of extremely high quality, [22].

[20], [21] and [22] summarized the use of MPEG-4 as the first internet video streaming compression technique. MPEG-4 was utilized effectively for streaming techniques but was of a

lower quality when compared to MPEG-1 and MPEG-2.

4.4 Motion Picture Experts Group – 7 (MPEG-7)

In 2001, MPEG-7 was approved as a video content description standard, because of some properties such as providing fast and effective searching in the digital content media. MPEG-7 stores metadata by means of the Extensible Markup Language (XML), [20].

One of the important benefits of MPEG-7 is the engagement in smart processing of video management software and video analytics applications by selecting the contents and events of such streams. And so, one of the most important applications of MPEG-7 is video surveillance applications stream, [20], [23].

4.5 Other Video Compression Standards

H.263 Compression Standard was developed in 1996; it adopts the Discrete Cosine Transform (DCT) method. It is significantly proposed for videoconferencing over Public Switched Telephone Network (PSTN) applications which have a very low bit rate, [20]. Nevertheless, using it in general digital video coding was not suitable, [20], [21].

H.264 Advanced Video Coding (H.264/AVC, or MPEG-4 part 10) was approved in 2003. It has several advantages such as low latency and high-quality video streaming, it is used in several applications such as mobile video conferencing and Ultra HDTV, [20]. However, H.264 is not appropriate for use in applications such as real-time video streaming due to the complexity of encoding and decoding processes, [24].

H.264/AVC Extension for Scalable Video Coding (H.264/SVC) was approved in 2007. It can efficiently handle different types of scalabilities; spatial, temporal, and quality scalabilities, it can employ one or different combinations of those three scalabilities based on the video communication, [25]. The previously mentioned scalabilities can provide extra functions such as gracious degradation in lossy transmission applications and bit rate, format, and power adaptation, [26]. So, H.264/SVC is more suitable for real-time (such as videoconferencing, remote medical consultation, and remote surgery) and live video (such as live YouTube, live news channels, and football matches) streaming systems than the former H.264/AVC standard, [24], [25].

[27] introduced a near real-time no-reference video quality assessment approach for videos encoded using the H.264/AVC codec. They trained a fully connected neural network using features extracted from both the bit-stream and pixel domains, along with corresponding subjective quality scores. The method demonstrated strong performance in terms of quality assessment.

High-Efficiency Video Coding (HEVC) is the latest compression standard, which was released in 2013. It outperforms all the previous compression standards; it keeps the level of perceptual video quality with a significant decrease in bit-rate by approximately 50%, [28]. Indeed, HEVC was designed to deal with high video resolution and high use of parallel processing architecture [28].

[29] proposed a practical method for High Dynamic Range (HDR) video tone-mapping, integrating two overlapping computational blocks within the HDR video tone-mapping and HEVC compression frameworks. This approach enables real-time HDR video tone-mapping, suitable for applications like video conferencing, remote medical consultations, and remote surgeries.

A dynamic model for predicting streaming video was proposed by combining various models into a forecasting ensemble, resulting in improved accuracy and reduced forecasting variance, [30].

[31] introduced the Display Stream Compression (DSC) standard, which allows for visually lossless video compression at a significantly lower hardware cost compared to H.264 and HEVC.

4.6 Video Compression techniques based on Artificial Intelligence (AI)

Artificial Intelligence (AI) and Machine Learning (ML) techniques can solve multi-dimensional issues that are difficult to model mathematically. (AI) plays a vital role in video compression since the pandemic of COVID19. This section provides a summary about some of AI-based video compression techniques.

[32] proposed a method to identify and compress regions in a frame without affecting the quality of remaining regions within the frame using AI. The proposed method performed well when compressing video in terms of vision accuracy in comparison to traditional video compression methods.

[33] developed a degraded model for compressed video images by utilizing a neural

network and the H.264 rate-distortion model. The neural network's predictive capabilities were used to enhance the quality of compressed images, improve quantization parameter prediction during video encoding, and facilitate video image reconstruction.

[34] proposed a method that mainly consists of downscale and upscale parts which allows downsizing of the transmitted frame and reconstruct it in a decoder with high accuracy. The decoder uses preprocessing before upscale operation and post-processing after frame size restoration. The proposed method works in conjunction with the classic video codec H.264/ A VC. The compression rate has been improved.

[35] proposed a method that utilizes a deep convolutional neural network (CNN) to approximate the inverse function of video compression, aiming to address blocking artifacts in block-based hybrid video coding caused by block-wise intra/inter prediction, transformation, and quantization. The method establishes an end-to-end mapping, where the CNN implicitly learns the relationship between the lossy and original frames under varying Quantization Parameters (QP).

CNN was implemented as a post-loop deblocking filter, with various CNN parameters explored for different QPs.

[36] proposed a machine learning-based approach for power and thermal management that dynamically determines the optimal encoder configuration and core frequency for video streams running on an MPSoC. The method uses information on frame compression, quality, performance, total power, and temperature to reduce the average temperature without compromising compression quality or increasing power consumption.

[37] proposed a CNN-based postprocessing technique, which has been incorporated into two advanced video coding standards: Versatile Video Coding (VVC) and AOMedia Video (AV1).

[38] introduced a Semantic-Aware (SA) video compression framework that independently and concurrently compresses the region-of-interest and region-out-of-interest in automotive camera video frames before transmitting them to the processing unit.

[39] introduced a new color video compression method that utilizes the greatest solution of a system of bilinear fuzzy relation equations to evaluate the similarity between video frames. Therefore, this AI compression technique achieved more accurate comparison and finding the similarity between different video frames (to reduce the similar video frames and enhance the video compression rate), So, this method is more suitable for comprising real-time video (such as video conferencing and WhatsApp video calling) and live video (such as live YouTube, live news channels and football matches) streaming systems.

5 Errors Correction Techniques for Enhancing Video Streaming Quality

Error correction techniques are employed for both error detection and correction. This section discusses the error correction methods used to improve the quality of video streaming over packet-switched and best-effort networks, such as the Internet.

5.1 Forward Error Correction (FEC) Techniques at the Application Layer

In FEC, the sender adds redundant (parity) bits to the source information symbols to enable the receiver to detect and then correct the bits in error on the fly as shown in Figure 2, [5]. FEC needs more bandwidth due to redundant bits, but it is a fast method for correction, so it is suitable for real-time and live video streaming applications, [40]. This work focuses on the following FEC techniques:

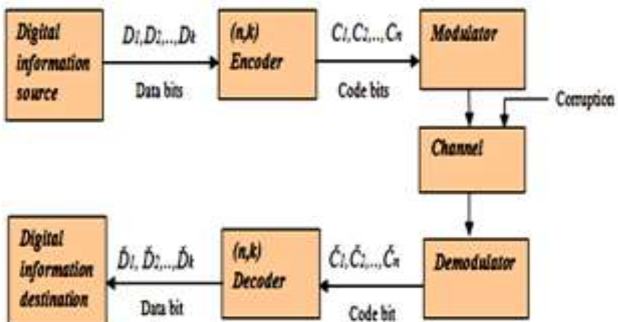

Fig. 2: General encoding and decoding processes in the FEC technique, [5]

5.1.1 Reed–Solomon Codes

Reed-Solomon code [41] is a cyclic error correction code which means that it can detect and correct multiple random errors. It is a linear block code of a specific length, and it can fix up to (n-k)/2 erroneous symbols, where n is the length of the code word, and k is the length of the source information symbol, [6].

5.1.2 Fountain Codes
Fountain codes [42] are a type of erasure rateless code, meaning they do not depend on a fixed code rate and can generate a flexible number of encoded symbols from the original source symbols. Fountain codes offer several advantages over traditional erasure codes like Reed-Solomon codes [43]: the encoder can produce encoded symbols on demand as needed, and the encoding and decoding processes are quicker than those of traditional erasure codes, making fountain codes well-suited for real-time video streaming.

5.1.3 Luby Transform (LT) Codes
LT codes represent the first practical implementation of erasure and rateless fountain codes, utilizing a straightforward algorithm based on the exclusive OR function, [44]. They offer several advantages, including the ability to generate several bits that closely approximate the minimal number of encoded symbols. However, one drawback is that LT codes are not systematic; the output symbols do not necessarily reproduce the input symbols, [45].

5.1.4 Raptor Code
Raptor codes [46] are the first type of fountain codes that utilize a linear time complexity algorithm for both encoding and decoding, which helps reduce end-to-end delay. Raptor codes can operate in both non-systematic and systematic modes.

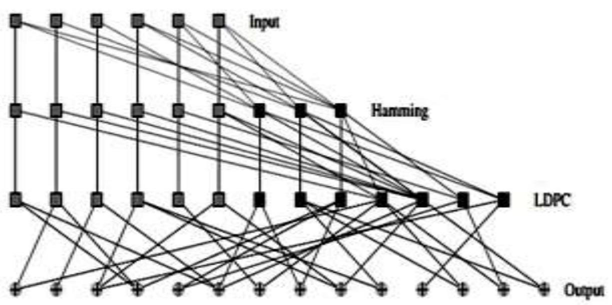

Fig. 3: One version of Raptor codes (Courtesy of [45])

The coding process consists of two stages: the first stage employs Hamming coding, while the second stage uses Low-Density Parity Check (LDPC) coding, as illustrated in Figure 3, [47].

5.2 Backward Error Correction (BEC) Techniques at the Application and Transport Layers
BEC is an error correction technique in which the receiver requests the sender to retransmit any corrupted or lost data, [4]. There are two methods for notifying the sender about the corrupted data: Positive Acknowledgment (ACK), and Negative Acknowledgment (NACK) methods (Figure 4 and Figure 5). Transmission Control Protocol (TCP) is a clear example of BEC techniques, [4].

One of the biggest disadvantages of BEC is the high average delay in the data-receiving process, therefore it is not suitable for real-time and live video streaming applications.

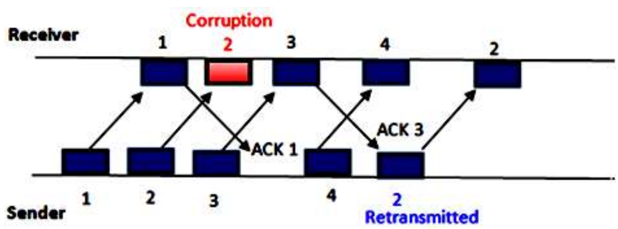

Fig. 4: BEC with positive Acknowledgment (ACK)

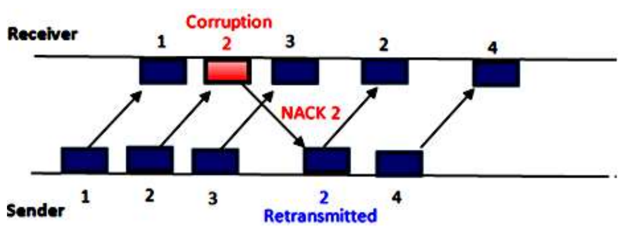

Fig. 5: BEC with negative Acknowledgment (NACK)

5.3 Other Error Correction Techniques
There are many error correction techniques that are listed next: TCP-Friendly Rate Control Protocol (TFRC) [48], Reliable UDP (RUDP) [49], Real-time Transport Protocol (RTP) [1], Bidirectional Transport Protocol (BTP) [49], Embedded Reliable Transport Protocol (ERT) [49], Datagram Congestion Control Protocol (DCCP) [49], Broadband Video Streaming (BVS) [49], Original Random Early Detection (RED) [50], Neighborhood RED (NRED), [51].

[52] proposed and outlined an architecture for de-identifying DICOM-RTV metadata in real-time before streaming it, along with DICOM-RTV video of the ongoing surgery, outside the medical facility.

6 Enhancing Video Streaming Quality using Forward Error Correction Codes with Feedback

This section highlights several techniques that have improved performance using Forward Error Correction (FEC) with feedback:

1) Enhancements in Video Streaming over the Internet: [53] proposed a combination of Multiple Description Reed-Solomon (MD-FEC) and Linear Increase Multiplicative Decrease with History (LIMD/H) to enhance video streaming quality over the Internet. However, this approach encountered challenges related to bandwidth overhead, packet loss, and delays.
2) Improving Quality of Distributed Video Streaming: [54] introduced an integrated solution to enhance video streaming quality in simple distributed networks regarding throughput, packet loss tolerance, and delay. Their method involved three senders and one receiver, successfully integrating the Receiver-Driven protocol with Reed-Solomon FEC coding.
3) Enhancing Peer-to-Peer Video Streaming Quality: In Peer-to-Peer (P2P) video streaming, where each participant functions as either a client or server, a hybrid approach combining Reed-Solomon FEC coding with a packet retransmission technique based on BEC was proposed to improve video streaming quality, [55].
4) Adaptive Live Unicast Video Streaming: Finally, an adaptive live unicast video streaming method utilizing rateless fountain codes and feedback was proposed in [56].

7 Enhancing Video Streaming Quality using Forward Error Correction Codes with Unequal Loss (or Error) Protection Techniques

Unequal Loss (or Error) Protection (ULP or UEP) is considered one of the most powerful techniques for enhancing the quality of live and real-time video streaming.

This section presents some techniques that enhance the quality of video streaming by implementing ULP or UEP.

7.1 Unequal Loss Protection Technique for Graceful Degradation of Single Still Image Quality over Packet Erasure Channels through Reed-Solomon Forward Error Correction Codes

This technique enhanced the transmission quality of a single still image over packet erasure channels using ULP. However, ULP methods that rely on Reed-Solomon with fixed-rate FEC codes are inadequate for transmitting live and real-time video streaming over packet erasure networks, which are characterized by dynamic and unpredictable conditions because the encoder of Reed-Solomon fixed-rate codes cannot generate the needed encoded symbols on demand, and it has slow encoding and decoding processes, [6], [43].

7.2 Applying Unequal Error Protection (UEP) with Rateless FEC Coding to MPEG Video Compression Standards

UEP with rateless FEC codes can enhance the quality of live and real-time video streaming. The techniques for protecting information symbols are categorized into two main classes: Least Important Bits (LIB) and Most Important Bits (MIB) symbols protection, [57], [58].

7.3 Enhancing Unicast and Multicast Video Streaming via using the recent static Unequal Error Protection with Rateless Luby Transform (LT) Codes

[43] introduced an efficient static UEP scheme using rateless LT codes for the scalable video coding (H.264/SVC) compression standard. This recent static UEP scheme was employed alongside LT rateless codes in multicast video streaming applications, accommodating heterogeneous receivers. The proposed technique improved the quality of real-time and live video streaming over packet erasure networks by achieving superior PSNR performance.

7.4 Enhancing Unicast and Multicast Video Streaming using Adaptive Unequal Error Protection with Rateless Luby Transform (LT) Codes

[7] proposed a heuristic-based algorithm to dynamically adapt the value of Expansion Factor (EF), the number of priority classes, and the value of Repeat Factor (RF) for each class according to the observed lost packets rate feedback from each

receiver as shown in Figure 6. The following steps summarize the heuristic-based algorithm assuming temporal scalability:

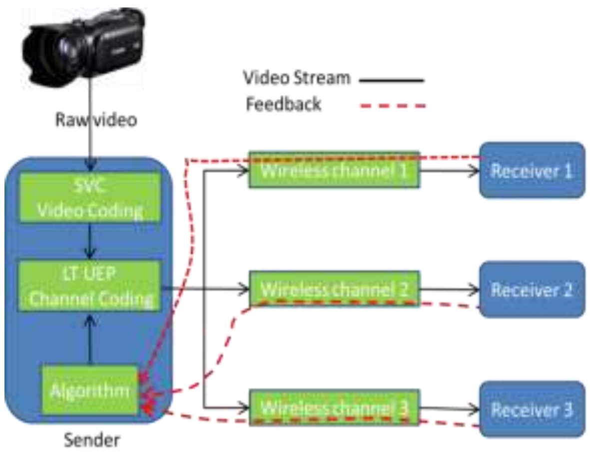

Fig. 6: The Adaptive UEP scheme (Courtesy of [7])

1) "If the average packet loss rate is lower than a threshold of 2%, then:
   a) The source symbols are divided into 3 classes: MIB (I frames), Average Important Bits (AIB) (P frames), and LIB (B frames)
   b) Set RF to 3 for MIB, RF to 2 for AIB, and RF to 1 for LIB symbols
   c) Set EF to 20

2) If the average packet loss rate is between (2% - 15%), then:
   a) The source symbols are divided into two classes: MIB (I and P frames) and LIB (B frames)
   b) Set RF to 14 for MIB and RF to 1 for LIB
   c) Set EF to 1

3) If the average packet loss rate is more than 15%, then:

   a) The source symbols are divided into 2 classes: MIB (I and P frames) and LIB (B frames)
   b) Set RF to 20 for MIB and RF to 1 for LIB
   c) Set EF to 1 for all MIB and LIB symbols to achieve the final virtual source block"

Actually, the proposed adaptive UEP scheme [7] outperformed all the previous techniques in the field of static UEP [43] and [59] in terms of average PSNR because of its dynamic nature (Figure 7). So, adaptive UEP technique is more suitable for the applications when high video quality is required.

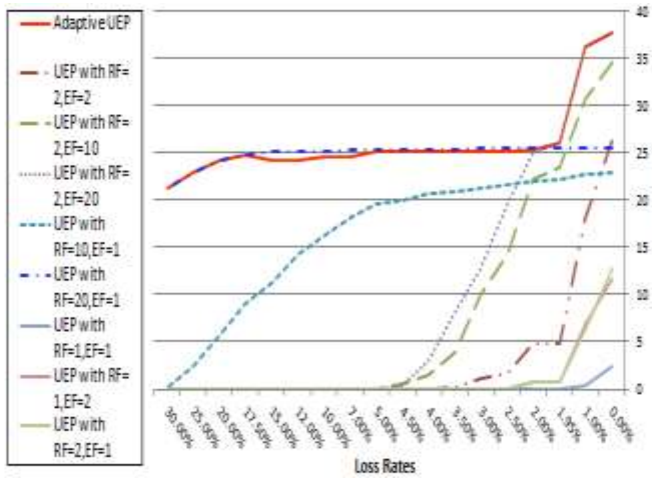

Fig.7: Average PSNR versus average observed packet loss rate using the adaptive UEP and static UEP techniques

8 The Layered Video Streaming

Video streaming consists of several layers; one base layer that provides the basic video quality (information), and many higher enhancement layers that help in refining the video quality, [24].

8.1 Layered Video Multicast with Retransmissions (LVMR) and Hierarchical Rate Control Techniques

LVMR is a protocol designed for handling video streaming over packet erasure networks with heterogeneous multicast receivers. [60] proposed that video streaming consists of multiple layers, including one base layer (representing the basic level of video quality) and several enhancement layers. Additionally, LVMR [60] introduced a BEC method that:

1) Improved the quality of reception within each layer by incorporating a method for retransmitting lost packets.
2) Adapted to network congestion and heterogeneity by implementing a hierarchical rate control technique to facilitate the addition and removal of video layers by the receiver.

On the other hand, BEC, Hierarchical Rate Control, and packet retransmission techniques used in [60] suffered from high average delay in the data receiving process (especially in lost packet retransmission parts). Therefore, [60] was too slow, and hence is not suitable for recent real-time video (such as video conferencing and WhatsApp video calling) and live video (such as live YouTube, live news channels, and football matches) streaming applications.

8.2 Layered Quality Adaptation for Internet Video Streaming by using Buffering Techniques at the Receiver End

Video streaming applications face two conflicting requirements: the application requirements (such as Quality of Service (QoS), delay, packet loss tolerance, and rate-based considerations) and the network requirements (including sharing, stability, and congestion management). [61] proposed an algorithm for dynamically adjusting the layered quality of congestion-controlled video streaming playback, along with a mechanism for optimizing buffer distribution.

8.3 H.264/SVC Layered Video Coding Compression Standard

H.264/AVC uses a single-layer coding technique, while H.264/SVC employs a layered coding approach. However, H.264/SVC offers the flexibility to encode a video into multiple layers. Overall, layered coding techniques, such as H.264/SVC, achieve greater coding efficiency compared to single-layer coding methods like H.264/AVC, [24].

9 Single and Multiple Paths Video Streaming

Single-path video streaming supposes that the video can be streamed over a single path among the existing paths between sender and receiver, which in turn reduces the scalability and deployment over the network (such as the Internet) and increases the congestion over the network. Multipath video streaming techniques solve the issues that occur by a Single path (achieving optimal QoS, increasing the capability to adapt to dynamic network conditions, and reducing the impact of packet losses) by sending packets over multiple paths.

There are three types of path multiple-path video streaming:

9.1 Multipath Video Streaming (Single Sender to Single Receiver)

Providing multiple paths up to a certain threshold value leads to distributing the load between many paths, which reduces the overall traffic and packet loss probability, which in turn enhances the QoS of delay-constrained video applications. But, when using the single shared path, packet loss will be increased and so affecting the QoS negatively, [62].

9.2 Multipath Video Streaming (Many Senders to Single Receiver)

When multiple sources send video streams to a single destination, the performance of video streaming can be improved regarding latency, scalability, fault tolerance, and packet loss by leveraging path diversity, [63].

[1] introduced an intelligent multipath video streaming approach involving multiple senders and a single receiver. In this method, a pre-stored MPEG1 video flow is divided into packets, with different packets taking alternate routes to reach the single receiver, resulting in high-quality video streaming, though it requires considerable bandwidth.

[64] proposed flexible Distributed Rateless FEC Codes with UEP scheme 1 [65], designed for multiple senders and a single receiver, where various data block lengths are assigned to different senders based on their priorities. However, multipath video streaming has some drawbacks, including potential costs and overheads, [66].

9.3 Multipath Video Streaming (Single Sender to Multiple Receivers)

A single sender can stream multiple copies of the same video to various receivers, a scenario commonly found in applications such as interactive and teleconferencing video over the Internet. [66] introduced the Multicast Heterogeneous Packet Flows (MHPF) technique, which facilitates multicast video streaming of heterogeneous data for small to medium multicast groups without breaking them down into component homogeneous data streams. MHPF transmits only the highest priority packets over the downstream paths, allowing each receiver to periodically generate packets over the upstream path. However, the system's performance may degrade as the number of receivers increases.

10 Video Streaming Over Wireless Networks

Wireless networks, including Wi-Fi, WiMAX, Bluetooth, and mobile phone systems, have certain disadvantages compared to their wired counterparts, such as lower data rates, unpredictable bandwidth, reliability issues, fading, interference, dynamic fluctuations in congestion levels, channel errors, lost packet rates, and PSNR. To achieve high throughput, these networks must fulfill the requirements of low delay and an acceptable lost packet rate for live and real-time video streaming transmission. In subsection 10.1, we will provide a brief overview of 3G, 4G, 5G, Wi-Fi, and WiMAX wireless networks. Subsequently, subsection 10.2, highlights some examples of how the quality of video streaming can be enhanced across different wireless networks.

10.1 Wireless Network Types

3G can handle all types of data effectively. 3G uses Wideband Code Division Multiple Access (WCDMA) radio technology, [67]. It provides a high-speed data rate of up to 384 kbps for moving vehicles and a minimum of 2 Mbps for stationary users or pedestrians, [68]. 4G mobile networks depend on the Orthogonal Frequency-Division Multiplexing (OFDM) technique. Therefore, 4G can provide up to 100 Mbps, furthermore, 4G offers better services (like HD video calling and streaming online gaming services), [69].

5G is designed to achieve up to 1 Gbps user-experienced data rate, [70]. 5G can support Full HD video calling, reliable and fast communication services, IOT, advancement in online secure banking, etc., [69].

Wi-Fi is 802.11 IEEE standard, and 802.11n is one of its standards, 802.11n has a maximum indoor range of 70 meters, a maximum outdoor range of 250 meters, and a maximum bandwidth of 40 MHz, [71].

WiMAX IEEE standard is 802.16, and 802.16m is one of its standards, 802.16m has a maximum range of 50-km, and an advanced Air Interface with data rates of 100 Mbps mobile and 1 Gbps fixed, [72].

It is useful to merge the advantages of Wi-Fi (802.11n) and WiMAX (802.16m) to produce a hybrid system WLAN-WMAN (or Wi-Fi-WiMAX) network. The hybrid network achieves wide network coverage, high data rates, and more economical gains for service providers because WLAN-WMAN increases the number of customers, [73].

Table 1 provides a detailed comparison between these networks.

Table 1. Comparison between different wireless technologies, [72]

Technology	Bluetooth	Wi-Fi – 802.11n	WiMAX
Application	Personal Area Network	Wireless LAN, Internet	Broadband Internet Connectivity
Typical Range	5 – 20 m	100m	50Km
Frequency Range	2.4 – 5 GHz	2.4 GHz	2 – 11 GHz
Data Rate	2 – 6 Mbps	54 – 108 Mbps	75 Mbps
Modulation	TDM	DSSS	QAM
Network	IP & P2P	IP & P2P	IP
IT Network Connectivity	No	YES	NO
Network Topology	Infrastructure	Infrastructure (Ad-hoc also Possible)	Infrastructure
Access Protocol	L2CAP/LMP	CSMA/CA	Request/Grant
Key Attributes	Less cost	Wider BW, Flexibility	Throughput, Coverage

10.1.1 Enhancing the Quality of Video Streaming over 3G Mobile Wireless Network

Due to limited bandwidth, 3G mobile wireless networks suffer from fading and interference which affect packets loss and PSNR negatively. 3G uses the Wideband Code Division Multiple Access (WCDMA) technique that supports the Multimedia Broadcast/Multicast Service (MBMS) method, MBMS uses systematic Raptor FEC, [74].

MBMS uses H.264/SVC video streaming compression technique rather than H.264/AVC to make it suitable for real-time and live video streaming, [24].

10.1.2 Enhancing the Quality of Video Streaming over 4G Networks

User equipment in 4G systems (such as mobile phones and tablets) are equipped with video capture capabilities, in addition to social media sites (e.g., Facebook, Instagram, and Snapchat), which produces huge growth in images and video streaming, this is tempered because of limited bandwidth, [75].

[76] proposed a scalable hierarchical wavelet decomposition-based layered video coding approach using a parallel TCP scheme for reliable video transmission over Multiple Inputs and Multiple Outputs (MIMO) wireless channels. This method aims to reduce delay and jitter in video streaming over 4G mobile networks.

10.1.3 Enhancing the Quality of Video Streaming over 5G Mobile Wireless Network

Traffic on 5G mobile wireless networks is driven by demanding real-time video streaming applications, such as mobile broadcasting, remote medical consultations, remote surgeries, and augmented reality, all of which require ultra-high-quality delivery, [77].

The 5G-QoE framework addresses Quality of Experience (QoE) modeling for ultra-high-definition (UHD) video streams in 5G networks, focusing on creating a QoE prediction model that is both sufficiently accurate and low in complexity for practical use, [78].

The integration of Scalable Video Coding (H.265/SVC) and Mobile Edge Computing (MEC) enhances real-time video streaming applications due to its reliability, scalability, and low-delay transmission, [77], [79]. [79] proposed a two-step approach using Lagrangian optimization under a dual pricing specification. This algorithm offers an explicit analytic solution and can be applied from a supply-demand pricing perspective, ultimately improving the quality of live video streaming (such as live YouTube broadcasts, news channels, and football matches) within the MEC-enabled cellular system.

Finally, [8] highlighted an adaptive real-time video streaming implementation over a 5G mobile wireless network, which is able to adjust to the prevailing uplink network conditions near real-time based on the gathered 5G network QoS information. Furthermore, [8] showed an extensive set of verification tests in an underground test environment using a moving vehicle and a private 5G standalone network. The results from these field tests presented the advantage of using video adaptation depending on 5G network QoS for reducing the uplink delay peaks in video streaming. Therefore, this technique is suitable for real-time video streaming (such as video conferencing, remote medical consultation, WhatsApp video calling and remote surgery).

10.1.4 Enhancing the Quality of Video Streaming over Wireless Local Area Networks (WLANs)

[80] proposed a method to enhance video streaming quality over WLANs by sending one acknowledgment for every ten received packets in a UDP-like TCP model, thereby reducing the acknowledgment frequency. This approach utilized the H.264/AVC single-layer coding technique for video compression.

[81] employed a Content-Adaptive Robust layered H.264/SVC video streaming compression technique over 802.11e WLANs, applying Unequal Error Protection (UEP) to manage video layers. They assigned high priority to the base layer to facilitate video reconstruction, effectively reducing the packet loss rate and improving PSNR over 802.11e WLANs. Furthermore, this technique achieved low decoding complexity, particularly in the Scalable Baseline profile and the lower levels of the Scalable High profile. Consequently, the method proposed in [81] using H.264/SVC is more suitable for real-time video streaming applications, such as video conferencing, remote medical consultations, and WhatsApp video calls, compared to the previous technique with the H.264/AVC compression standard, [80].

In high-quality real-time video streaming on YouTube, network administrators encountered new challenges in managing the network and meeting expected Quality of Experience (QoE) standards. To address this, [82] introduced a technique that dynamically adjusts IEEE 802.11 parameters to improve network conditions and, as a result, enhance QoE. This method relies on Double Deep Q-Learning (DDQL), enabling the WLAN controller to learn the optimal configuration of access points (APs)—in terms of transmission power and channel number—that yields the best QoE. The proposal in [82] improved video QoE by up to 91% in the best-case scenario compared to three baseline methods, while also balancing QoE among clients, enhancing fairness by up to 115% compared to the baselines.

10.1.5 Enhancing the Quality of Video Streaming over WiMAX using Inter-Frame Retransmission

[49] introduced the Inter-frame Retransmission (IR) protocol to address the issue of lost packets over WiMAX. This protocol sends a single negative acknowledgment (NACK) packet to notify the

sender of all lost packets within a specific video frame (Figure 8). The IR protocol resulted in minimal average delays for packets and frames.

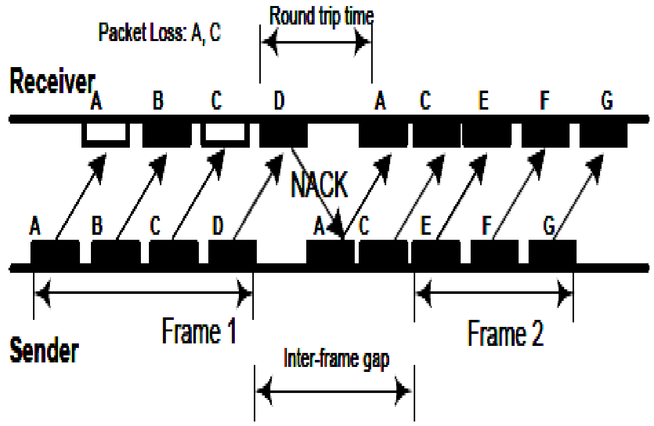

Fig. 8: The Inter-frame Retransmission (IR) NACK (Courtesy of [49])

In Quick Response (QR), the receiver has the option to send either a negative acknowledgment (NACK) packet for each lost packet, which is appropriate when only one packet is missing in the frame, or a single NACK packet for an entire frame, which consumes more bandwidth. In contrast, the Inter-frame Retransmission (IR) receiver only sends one NACK for all lost packets within a single frame. As a result, IR produces fewer NACK packets than QR, which benefits both the delay and the number of interruptions.

Some packets are prioritized in NACK packets are sent to them only which decreases packet and frame delays but increases packet loss.

If some packets are lost in a frame, a Congestion Delay (CD) can be added to make sure that the current frame arrives before the next one (Figure 9). Therefore, adding the CD technique reduces the average delays, which decreases the cumulative throughput. However, prioritizing IR with CD leads to enhancing the quality of video streaming (or PSNR) over WiMAX wireless networks (Figure 10).

Finally, [9] proposed the design of a high-efficiency deep learning based low-BER (Bit Error Rate) real-time Video streaming model for high-noise IEEE 802.16 WiMAX wireless network. The model proposed in [9] initially employs a Long-Short-Term Memory (LSTM) block to estimate frame-level features, followed using an Orthogonal Frequency Division Multiple Access (OFDMA) modulation platform for the transmission and reception of the processed video frames. Video frames undergo pre-processing with Dual Neural Networks, which aid in estimating differential frame information sets. This estimation of differential frame information enhances streaming speed, allowing for an increased number of frames transmitted per second, thereby improving streaming performance across various video types. This makes the model particularly beneficial for a wide range of real-time video streaming applications, including video conferencing, remote medical consultations, WhatsApp video calls, and remote surgeries.

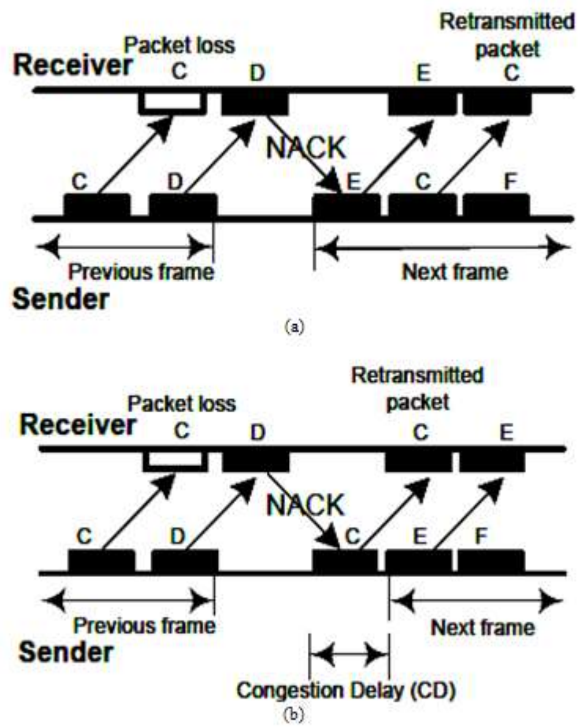

Fig. 9: Comparison between (a) IR without CD and (b) IR with CD (Courtesy of [49])

11 Packet Erasure Networks /Packet Erasure Channels

Erasure packets pose a significant challenge, prompting several research efforts to address this issue:

[83] identified optimal rates for erasure and channel codes to minimize the transmit power needed for a specific message error probability. [84] explored the achievable throughput of batched temporal network coding in multi-hop erasure channels. [85] examined a cache-aided broadcast network where a server delivers content to a group of receivers over a packet erasure broadcast channel.

[86] focused on a cache-enabled K-user broadcast erasure packet channel, in which a server with a library of files aims to deliver a requested file to each user equipped with limited cache memory. [87], [88] investigated 1-to-2 broadcast packet erasure channels with causal ACK, motivated by practical downlink access point networks.

12 Conclusion and Future Work

This work provides a detailed review of related topics to real-time and live video streaming including compression and error detection and correction techniques over several types of wireless networks, providing brief comparisons between different techniques. From subsection 4.5, it is noticeable that HEVC leads to a significant reduction in bitrate by approximately 50% while keeping the same level of perceptual video quality as in the former standards. Furthermore, DSC lossless video compression standard achieved lower hardware costs than H.264 and HEVC.

Sub-section 4.6 highlighted some current video compression techniques that identified and compressed regions in a frame without affecting the quality of remaining regions within the frame using Artificial intelligence (AI) techniques. Furthermore, in sub-section 4.6, we discussed an SA video compression framework that separately and simultaneously compressed the region-of-interest and region-out-of-interest of automotive camera video frames prior to transmission to the processing unit. Additionally, in the same sub-section, we introduced an innovative color video compression method that utilized the optimal solution of a system of bilinear fuzzy relation equations to evaluate the similarity between video frames. Therefore, these AI compression techniques achieved more accurate comparison and finding the similarity between different video frames (to reduce the similar video frames and enhance the video compression rate), so these new video compression techniques outperformed the previous old video compression techniques and achieved better compression rates. To conclude these AI compression techniques (presented in sub-section 4.6) are more suitable for real-time video (such as video conferencing, remote medical consultation, and WhatsApp video calling) and live video (such as live YouTube, live news channels, and football matches) streaming applications.

Sub-section 5.1.4 presented the Raptor Forward Error Correction (FEC) code, the first type of fountain code featuring a linear time complexity algorithm for encoding and decoding, which helps to minimize end-to-end delay. From the analysis in sub-section 7.3, it is evident that the adaptive UEP technique significantly outperforms the static UEP technique in terms of average PSNR due to its dynamic nature. Consequently, the adaptive UEP technique is more suitable for applications that demand high video quality.

This work highlighted the enhancement of the quality of video streaming over 3G, 4G, 5G, Wi-Fi, and WiMAX wireless networks. sub-section 10.2.3 showed some systems that improve the quality of live video streaming in the MEC-enabled 5G cellular system. Also, sub-section 10.2.4 showed a technique which depended on double deep Q-Learning (DDQL), then the WLANs controller can learn the configuration of the access points (APs) that enhanced the QoE. Furthermore, it can be concluded (from sub-section 10.2.5) that the prioritized IR with CD helps in decreasing the average delay and preserving the prioritized frames; Therefore, it leads to enhancing the video streaming quality (or PSNR) over the WiMAX wireless network. Finally, from sub-section 10.2.5 it can be concluded that the proposed design of a high-efficiency deep learning-based low-BER (Bit Error Rate) real-time Video streaming model led to improving real-time video streaming speed for different real-time video applications (such as video conferencing, remote medical consultation, WhatsApp video calling, and remote surgery) over high-noise IEEE 802.16 WiMAX wireless network.

This work effectively presents clear and comprehensive insights into the current and most significant techniques employed to enhance the quality of real-time and live video streaming over both wired and wireless networks. As such, it can serve as a valuable resource for students and researchers in their ongoing studies.

Video streaming's security is one of the most important research domains nowadays, therefore in future work, we will review live and real-time video streaming's security over advanced wireless networks (e.g., 5G and 6G).

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