Media Streaming Technologies: Current and Future Trends

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Abstract:
Since the introduction of H.261 video coding standard in 1988, media streaming has attracted tremendous attention of the scientific community, and has been constantly evolving ever since. In the 90's of the past millennium, media steaming started as HTTP progressive download, but by the end of the decade there were already dedicated media streaming servers based on the Macromedia Flash player and RTMP protocol. In the first decade of the 21-st century we witnessed the emergence of media streaming platforms such as YouTube, Amazon Prime Video and Netflix, while the second decade marked the return of HTTP media streaming with the rise of streaming protocols for dynamic adaptive streaming over HTTP such as Apple's HLS, Microsoft's Smooth Streaming, Adobe's HDS and MPEG's DASH. Furthermore, this decade marked the introduction of Media Source Extensions, Encrypted Media Extensions, WebRTC and some other technologies that revolutionized media delivery over streaming networks. On the verge of the third decade of this century some novel trends are on the rise. In this paper, we elaborate the current technologies and recent advances in media streaming and we conclude the paper with emerging trends in media delivery over streaming networks.

Keywords:
Audio/video coding, media streaming protocols

1. Introduction

Audiovisual media has been fascinating humanity since their inception, and media technologies continued to steadily develop in time. With the invention of television, by the middle of the 20th century many people could enjoy multimedia from the comfort of their own homes. Today, computers, tablets, and smart phones are the main devices on which multimedia resources are interactively presented and the basic distribution technology is media streaming over IP networks.

The beginnings of media streaming are marked with the publication of two recommendations by the ITU-T (International Telecommunication Union – Telecommunication Standardization Sector), formerly known as CCITT (Comité Consultatif International Télégraphique et Téléphonique), both published in November 1988. The two recommendations are Rec. I.120 (Integrated Services Digital Networks - ISDN) [1] and Rec. H.261 (Codec for Audiovisual Services) [2].

H.261 was designed for media transfer over ISDN lines with bitrates that are multiples of 64 kbps. The coding algorithm works with bitrate range up to 2 Mbps, supports CIF (352x288) and QCIF (176x144) resolutions and uses Y'CbCr color model with 4:2:0 Chroma subsampling. The most interesting part of the H.261 video coding algorithm was that the foundations laid here, i.e. the use of Discrete Cosine Transform (DCT), Quantization, Zig-Zag scanning and Entropy coding are techniques used in all subsequent video coding standards such as MPEG-1 Part 2, H.262/MPEG-2 Part 2, H.263, MPEG-4 Part 2, H.264/MPEG-4 Part 10/AVC, H.265/HEVC, H.266/VVC.

The development of contemporary media streaming is not solely based on achievements in audio/video coding, but several other technologies as well, such as media streaming protocols, media player technologies, digital rights management and content delivery networks. In this paper we concentrate on audio/video coding (compression) and media streaming protocols only, we elaborate both technologies and we present the historical and latest achievements.

The remaining of this paper is organized as follows. Section 2 presents historical overview of audio/video coding standards and non-standard formats relevant for video streaming. In section 3 we elaborate the streaming protocols used in the past, the present and some promising trends. In section 4 we present the state of the art research in media streaming. Section 5 discusses some new trends that should be expected in near future, while section 6 concludes the paper with summary of the research.

2. Audio/Video Coding

The most prominent bodies that made a huge impact on techniques for audio/video coding are Video Coding Expert Group (VCEG), formed in 1984 by the ITU-T, and MPEG (Motion Picture Experts Group) formed in 1988 by the International Organization for Standardization (ISO) and the International Electro-technical Commission (IEC). Both, VCEG and MPEG partnered in 1991 to produce common new coding standards, while from 2001 to 2009 worked as a single entity, entitled Joint Video Team (JVT), to work on new audio/video coding algorithms. In 2010, VCEG and MPEG formed a new team called Joint Collaborative Team on Video Coding (JCT-VC), while in 2017 they formed the Joint Video Experts Team (JVET), where each new team had the same goals, i.e. to enhance the compression of video and audio over the previous published standard.

As we mentioned in the introduction, the first standard for video (and audio) coding was ITU-T Rec. H.261, published in 1988. It was the first of the H.26x family of standards, designed for video communications over ISDN lines. The maximum supported video resolution was only 352x288 pixels, but H.261 payed a huge role in the development of the next generations of video codecs. The audio codec used with H.261 was G.722, a codec intended for voice coding only.

MPEG-1 (ISO/IEC 11172) [3] is a standard published by the MPEG group in 1991 with a goal of video transfer over T1/E1 links and Video CDs, with bitrates of 1.5 Mbps. MPEG-1 consists of five parts, among which the most relevant for this paper are Part 2 – the video codec, and Part 3 – the audio codec. The history showed that the most significant contribution of this standard was the Audio Layer 3 of Part 3, popularly known as MP3.

The following standard in the line is MPEG-2 (ISO/IEC 13818) [4], also known as ITU-T Rec. H.262 [5]. It was jointly published by the two aforementioned organizations in 1995 with a goal of providing a standard for Standard Definition Digital Television (SDTV). Quickly after its publication MPEG-2 Part 2 / H.262 became the base for digital television standards such as ATSC (Advanced Television Systems Committee) and DVB (Digital Video Broadcasting), both for SDTV and high definition television (HDTV). Maybe the success of MPEG-2 is best described by the demise of MPEG-3, which was started to become standard for HDTV, but quickly stopped after the realization that MPEG-2 possessed all the potential to be used for HDTV. Furthermore, MPEG-2 Part 7 defines the new audio codes, named Advanced Audio Coding (AAC), later redefined (updated) as MPEG-4 Part 3. It was an improvement over the former MP3 standard, but remained fairly unpopular for many years after its publication.

The JVT in 2003 published perhaps the most important standard for video streaming to date. It is MPEG-4 Part 10 (ISO/IEC 14496) [6] / ITU-T Rec. H.264 [7], or otherwise known as Advanced Video Coding (AVC). AVC reaches identical video quality as MPEG-2 / H.262 with one third of the bitrate. Compared to MPEG-4 Part 2, for the same bitrate and visual quality, AVC encodes in four times higher resolution. Besides the greater efficiency, AVC provides greater visual quality when its compression limits are approached and graciously loses picture quality.

In 2013, the JCT-VC published the ITU-T Rec. H.265 [8] / MPEG-H Part 2 (ISO/IEC 23008) [9] video codec, known as High Efficiency Video Coding (HEVC). HEVC was published with an aim to provide 50% bitrate reduction over AVC, for the same visual quality.

One of the most recent video coding standards in this line, first published in 2020, is MPEG-I Part 3 (ISO/IEC 23090) [10] / ITU-T Rec. H.266 [11], also known as Versatile Video Coding (VVT). It was designed with two primary goals. First, to specify a video codec with compression capabilities that are substantially beyond those of the prior generations of such video coding standards, and second, to be highly versatile for effective use in a broadened range of applications.

MPEG group of ISO/IEC, in the last couple of years, published two more coding standards under the MPEG-5 (ISO/IEC 23094) [12] name. Part 1 - Essential Video Coding (EVC), published in 2020 and Part 2 - Low Complexity Enhancement Video Coding (LCEVC), published in 2021. The goal of MPEG-5 EVC is to provide a standardized video coding solution for business needs in some use cases such as video streaming. The MPEG-5 LCEVC specification defines two component streams, a base stream decodable by a hardware, and an enhancement stream suitable for software processing, and is intended for on demand and live streaming applications.

Two more, non-standard, video codecs compete with the aforementioned standards for video streaming applications. Video Project 9 (VP9) is a video compression format developed by Google and published in 2013. It is based on the previous similar codecs developed by On2 Technologies (formerly The Duck Company), which Google acquired in 2010. VP9 competes with high efficient codecs such as HEVC, and has the advantage for its royalty-free license and of being open and supported in modern Web browsers.

In 2015, Microsoft, Google, Amazon, Netflix, Intel, Mozilla and Cisco established the Alliance for Open Media (AOMedia) with an aim to create video standards that can serve as royalty-free alternatives to the dominant standards of MPEG and VCEG. In 2018, AOMedia released AV1 [14], an open and royalty-free video coding format, initially designed for video streaming applications. AV1 is based on Google’s planned VP10 project that was aimed as an improvement over VP9 codec.

Regarding the audio compression, the newest standard is Opus [15], initially developed by the Xiph.Org Foundation and in 2012 standardized by Internet Engineering Task Force (IETF). Opus is designed to efficiently encode speech and general audio in a single format, and it is said to offer higher-quality than other standard audio formats at any bitrate, including MP3 and AAC.

Another audio format that deserves attention is Vorbis [16], developed in 2000 by the Xiph.Org Foundation as well, which offers similar audio quality as MP3 and AAC. Vorbis has been used for streaming by some national radio stations, such as Deutschlandradio, Radio New Zealand and Absolute Radio, as well as by Spotify audio streaming service.

3. Media Streaming Protocols

The delivery of audio and video content via streaming requires the use of certain streaming protocols. These protocols represent specific standardized rules and methods that break up media files into smaller pieces, thus deliver that media to the end users as live content or on demand.

The oldest protocol that was used for media streaming was the Hypertext Transfer Protocol (HTTP) [17]. HTTP manages the communication between the web browser and the web server for the delivery of HTML pages, including images and other types of files. These early experimental efforts for media delivery over HTTP were not satisfactory for several reasons, among which the more important were the extremely limited bitrates of 28/56 Kbps with the dial-up connections in the 1990’s. In that time, Apple tried to promote the paradigm of HTTP progressive download, where the media presentation starts before the media file is fully downloaded, but this concept was consuming much of the available server bandwidth, because the media files were sent “as soon as possible”.

In the late 1990’s we witnessed the rise of dedicated media streaming protocols. Real-Time Messaging Protocol (RTMP) [18], developed by Macromedia, the company that was acquired by Adobe in 2005, was one of the mostly used streaming protocols, which is still in use today. Other streaming protocols that made an impact were Real Time Streaming Protocol (RTSP) [19], developed by Real Networks, Netscape and the University of Columbia, and standardized by the IETF in 1998, and the Microsoft Media Server (MMS) [20] protocol, also developed in 1998. These protocols required dedicated media streaming servers that worked together with HTTP servers to accomplish media selection and delivery. This enabled overcoming of some serious shortcomings of HTTP when used for streaming, such as the lack of control over the media presentation and the use of TCP instead of UDP for media transport. However, the dedicated streaming protocols were not perfect. The most common problems included the possibility for Firewall blockage of media packets, the inability to utilize the common caching mechanism at the Internet Service Providers (ISP) and the costs to run a separate media server.

Due to the aforementioned shortcomings, after more than a decade of dominancy of dedicated streaming protocols, the streaming community went back to the HTTP with novel streaming paradigms, commonly referred to as HTTP Dynamic Adaptive Streaming. It's a concept that addressed the previous problems, such as media presentation control over HTTP, and additionally enabled to adaptively switch among multiple streams with different bitrates. Four protocols of such technology are frequently used for streaming. Apple's HTTP Live Streaming (HLS) [21] and Adobe's HTTP Dynamic Streaming (HDS) [22], both developed in 2009, Microsoft's Smooth Streaming [23], developed in 2010 and MPEG-DASH [24], published as a standard by the ISO/IEC in 2012.

More recent protocols for media streaming are Web Real-Time Communication (WebRTC) [25, 26], Secure Reliable Transport (SRT) [27] and High Efficiency Stream Protocol (HESP) [28].

WebRTC protocol supports real-time media streaming for bi-directional communication. It can be used for ingestion and distribution with an end-to-end latency between 300ms - 600ms. The protocol was developed by Google and released in 2011. WebRTC specifications have been published by the World Wide Web Consortium (W3C) in December 2020 [25] and IETF in 2021 [26]. WebRTC has become the standard for real-time video communication on the web. The components that WebRTC is based on are accessible via a JavaScript API maintained by the W3C and the IETF, allowing users to live stream directly to a web browser without installation of any third-party tool.

SRT is an open source media streaming protocol that offers security, reliability and compatibility of high-quality and low-latency live video over the Internet. It was initially developed by Haivision in 2013, but released as open source protocol in 2017. SRT is maintained by the SRT Alliance consisting of many members, among which are Microsoft, Panasonic, Sony, Google Cloud, Alibaba Cloud, Canon etc. SRT is capable of delivering high-quality media streaming even when the network conditions are erratic. It also allows its use with any audio and video codec.

HESP is an adaptive HTTP based video streaming protocol, projected to bring superior Quality of Experience (QoE) for online viewers, while reducing the costs for scaling media delivery of up to 20%. HESP enables sub-second end-to-end latency as low as 400ms, and with zapping, start-up and seeking times well under 100ms, and it is claimed it achieves experiences better than the existing broadcast solutions. HESP protocol is developed by the THEO Technologies and maintained by the HESP Alliance. HESP protocol was first published in 2020 and submitted for standardization at IETF on May 20, 2021. Current active Internet draft is HESP version 2 from May 13, 2022 [28].

4. Latest Research in Media Streaming

Hongzi Mao, Ravi Netravali, Mohammad Alizadeh [29] proposed a new system that generates adaptive bitrate (ABR) algorithms, as an enhancement to MPEG-DASH media streaming. Their system, named Pensieve, is designed to train a neural network model that selects bitrates for future DASH media chunks. It does so by learning to make ABR algorithm decisions based on the resulting performance on past decisions. In experimental comparisons the authors claim that their ABR algorithm outperforms other algorithms, with improvements in average QoE of 12%–25%.

In similar research, to advance HTTP adaptive streaming, Christos G. Bampis et al. [30] developed a database, which contains subjective QoE responses to various design dimensions, such as bitrate adaptation algorithms, network conditions and video content. Using their database, they studied the effects of multiple streaming dimensions on user experience, evaluated video quality and QoE models, and analyzed their strengths and weaknesses. Their main conclusions were that average video quality and re-buffering duration were the most important factors contributing to accurate overall QoE prediction, but there is significant room for improvement of continuous-time QoE models.

Another research that deals with QoS database for adaptive media streaming is the work of Zhengfang Duanmu, Abdul Rehman and Zhou Wang [31]. They also concentrate on ABR algorithms, because these algorithms are not defined within the HTTP adaptive streaming standards, but deliberately left open for optimization. Testing different ABR algorithms has proven that no single algorithm performs best for all network profiles, which suggests that there is still room for improvements. In particular, proper combination of the ideas used in different ABR algorithms has the potential to further improve the performance.

Alireza Erfanian et al. in [32] introduced software-defined networking (SDN) concept and network function virtualization (NFV) technologies to create new, cost-aware, video streaming approach in order to provide AVC-based live streaming services. The video distribution is realized via DASH protocol, where clients' requests are collected at the edge of the network and sent to the SDN controller for determination of an optimal multicast tree for video transfer. Based on the performance results, the authors claim that their concept surpasses other AVC-based multicast and unicast approaches in terms of cost and resource utilization.

Mohammad Hosseini and Viswanathan Swaminathan [33] propose a dynamic view-aware adaptation technique for 360 Virtual Reality (VR) video streaming. In this technique, videos are spatially divided into multiple tiles and encoded using MPEG-DASH with Spatial Relationship Description (SRD) feature, to describe the spatial relationship of tiles in the 360-degree space, and prioritize the tiles in the Field of View (FoV). Their initial evaluation results revealed that bandwidth savings were up to 72% on 360 VR video streaming with minor negative quality impacts, compared to the baseline scenario when no adaptations are applied.

Bo Han, Feng Qian, Lusheng Ji and Vijay Gopalakrishnan [34] proposed a multipath framework for video streaming with awareness of network interface preferences from the users. Their overall goal was to enhance multipath TCP to support adaptive video streaming under user-specified interface preferences. They use HTTP adaptive streaming because of its ability to use any video codec. Their experiments at 33 locations in three U.S. states suggest that the framework is very effective, with reduction of cellular usage by up to 99% and radio energy consumption by up to 85%, with negligible degradation of QoE, compared to the off-the-shelf multipath TCP.

Matteo Gadaleta et al. [35] presented a framework that combines deep learning and reinforcement learning techniques to optimize the QoE in DASH streaming. The authors claim that their D-DASH algorithm performed better than several of the most popular adaptation approaches from the literature, maintaining a high video quality without paying a significant cost, either in terms of re-buffering events or stability of the quality.

Tianchi Huang et al. [36] proposed a video quality-aware ABR approach that improves the learning-based methods by tackling the low sample efficiency and lack of awareness of the video quality information. Their test results reveal that the ABR approach, named Comyco, outperforms previously proposed methods, with improvements on average QoE of 7.5% to 16.79%. Their most important claim is that Comyco surpasses the state-of-the-art approach Pensieve [29] by 7.37% on average video quality under the same re-buffering time.

Abbas Mehrabi, Matti Siekkinen, and Antti Ylä-Jääski [37] present an optimized solution for network assisted adaptation, specifically targeted to mobile streaming in multi-access edge computing environments. With this research, the authors intended to demonstrate the efficiency of their solution and to quantify the benefits of network-assisted adaptation over the client-based approaches in mobile edge computing scenarios, because the majority of approaches used today for bitrate adaptations are client based. The results from their simulations have shown that the network assisted adaptation outperforms the client-based DASH adaptations in some metrics, particularly in situations when the achievable throughput is moderately high or the link quality of the mobile clients does not differ from each other substantially.

Lu Liu et al. [38] propose an integration of wireless multimedia systems and deep learning. They decompose a wireless multimedia system into three components, i.e. end-users, network and servers, and present several potential topics to embrace deep learning techniques. Furthermore, they present deep learning based Quality of Service (QoS) / QoE prediction and bitrate adjustment as two case-studies. They claim that they achieved improvement of QoS compared to the baseline algorithm. They also show that the perceived video QoE average bitrate, re-buffering time and bitrate variations can be significantly improved.

5. Emerging Trends in Media Streaming

One of the emerging trends in video streaming is volumetric video streaming. Volumetric videos are truly three dimensional videos, allowing six degrees of freedom movement for their viewers during playback. Such flexibility enables numerous applications in entertainment, healthcare, education, etc., but volumetric video streaming is extremely bandwidth-intensive. One of the most recent research effort in this area is the work of Jie Li et al. [39], where they propose a hybrid visual saliency and hierarchical clustering empowered 3D tiling scheme that better matches the user's field of view, and they build a QoE model considering the volumetric video features as the optimization objective. The test results of their prototype system reveal that the proposed tiling and transmission scheme performs significantly better than the comparison schemes.

Another contemporary area of research in media streaming technologies is Video Streaming over Vehicular Ad Hoc Networks (VANET). VANET and Internet of Things (IoT) are considered as key elements in Intelligent Transportation Systems (ITS). For example, Debanjan Roy Chowdhury, Sukumar Nandi, and Diganta Goswami [40] explore video streaming solution with gateway minimization in constrained time data delivery to end-users. They propose network layer cooperation instead of application layer cooperation for gateway-client association. Their novel multicast protocol is specialized in streaming data distribution for dynamic scenarios, using either topology based or position based routing. Its performance is compared to the existing protocols and concluded that their protocol is most effective in service cost minimization while it is able to achieve competitive QoE performance.

Quite interesting trends in media streaming are WebRTC and streaming for IoT. In this manner, Robert R. Chodorek, Agnieszka Chodorek, and Krzysztof Wajda [41] focus on the adaptability of dual-stack WebRTC-based IoT transmissions. They use the capabilities of full-stack WebRTC transmissions, which merges media and non-media streams and flows, the capability to enable natural integration of different types of data in one session, the common cryptographical protection of the session, and the possibility for multi-platform applications development. The authors claim that the full-stack WebRTC communication assures good adaptability to network circumstances.

Furthermore, Gang Shen et al. [42] present a 360 immersive media solution using Intel-incubated Open WebRTC Toolkit (OWT) and edge computing platforms, while allowing media ingestion over 5G networks from multiple cameras, media control and 360 media distribution over 5G networks.

Aoyang Zhang et al. [43] propose an edge-assisted adaptive video streaming solution, which integrates super-resolution and edge caching to improve users' QoE. The authors designed a novel edge-based ABR algorithm that makes bitrate and video chunk source decisions by considering network conditions, QoE objectives, and edge resource availability. The solution, named VISCA, utilizes super-resolution to enhance the cached low-quality video at the edge. A novel cache strategy is also adopted to maximize caching efficiency. The prototype performance results reveal that compared to the existing video streaming solutions, VISCA improves video quality by 28.2% to 251.2% and reduces re-buffering time by 16.1% to 95.6% in all considered scenarios.

Miran Taha et al. [44] explore the possibilities to stream Ultra High Definition (UHD) video to users over wireless networks. They propose a smart algorithm for video streaming services to optimize assessing and managing the QoE of clients. The proposed algorithm includes two approaches. First, using the machine-learning model to predict QoE and second, according to the QoE prediction, the algorithm manages the video quality of the end-users by offering better video quality. As a result, the authors claim that the proposed algorithm outperforms previously proposed methods for predicting and managing QoE of streaming video over wireless networks.

6. Conclusion

In the last few decades, media streaming has gained huge attention from the scientific community, which is driven by the increasing use of media streaming services by more and more people every year. Several reports from market research entities, such as Grand View Research [45], Precedence Research [46] and Research and Markets [47], project constant growth of streaming market size and revenues between 2022 and 2030, with Compound Annual Growth Rate (CAGR) between 18.45% and 21.3%. This extensive market is enabled by the latest technologies for audio/video coding and streaming, which we reviewed in this paper. Regarding video coding for streaming, the most used codecs at present are MPEG-4 Part 10/ H.264 and VP9. AV1 is on the rise and H.265 is not yet sufficiently supported. As for audio coding, Opus codec appears to be mostly deployed in media streaming applications, followed by AAC and Vorbis. Promising video codecs for near future are

VVC and AV1, with better forecasted implementation of AV1 because of its royalty free licensing. As for audio coding for streaming applications, the Opus codec is still the dominant codec with a prospective to remain as such in the near future.

Concerning the streaming protocols, HTTP adaptive streaming is the de facto standard at present, and the latest research is mostly concentrated on the development of ABR algorithms. In the near future it is expected that WebRTC is going to receive increasing number of implementations, along with immersive media distribution, streaming for IoT devices and delivery of UHD video.

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