Journal of Innovative Technology Convergence
Vol. 5, No. 1 June 2023, pp. 21-30

Challenges and Opportunities in QoS Enhancement for Internet-Based Multimedia Communication

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Abstract: Advancements in communication technologies have fueled the rapid expansion of the multimedia content delivery market, encompassing various media types such as images, audio, and video. Personal communication, computing, broadcasting, and entertainment heavily rely on multimedia content, which is transmitted through diverse communication and network technologies to reach a wide range of devices. QoS requirements for multimedia services are determined by parameters including bandwidth flexibility, low end-to-end delay, minimal delay variation, and acceptable error rates. These parameters are interconnected and vary depending on the type of multimedia application. However, the QoS needs of multimedia communications cannot be fully met by the Internet's basic infrastructure. Mechanisms for ensuring QoS guarantees over the Internet have been developed through extensive research and development. This study examines developments, difficulties, and strategies for improving the transmission of multimedia content over the Internet, providing an overview of the methodologies and strategies used to satisfy QoS criteria in multimedia communications.

Keywords: Multimedia Communication, Quality of Service (QoS), Integrated services, Differentiated services, Multimedia services

1. Introduction

In recent years, significant advancements in communication technologies have led to a rapidly expanding market for multimedia content delivery like images, audio, and video. It's evident that personal communication, computing, broadcasting, and entertainment now heavily rely on multimedia content, which is transmitted through different communication and network technologies to reach a wide range of devices [1]. With the increasing deployment and widespread use of the internet in recent years, there has been a growing desire among people to utilize it not only for traditional data communications but also for multimedia communication. Quality of Service (QoS) is crucial for delivering satisfactory multimedia services while transmitting multimedia content over the Internet [2][3].

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Received [March 12, 2023]; Revised [April 27, 2023]; Accepted [May 25, 2023]

Published by InnoCon Publishing
ISSN 2704-4440

QoS requirements for multimedia services are typically determined by four key parameters:

The need for flexible and adjustable bandwidth to accommodate varying multimedia content requirements.

Minimizing the time it takes for data packets to travel from the source to the destination, ensuring real-time and responsive multimedia experiences.

Reducing fluctuations in packet delivery time, also known as jitter, to maintain smooth and consistent multimedia playback.

Tolerating a certain level of data loss or errors without initiating retransmission, as the additional delay caused by retransmission would be unacceptable for multimedia applications.

The above-mentioned QoS parameters are closely related to each other and dependent on the type of multimedia application. The multimedia applications have been classified into three categories based on the QoS requirements: *i.e.*, conversational applications, broadcasting applications, and content-on-demand applications [4]. In QoS requirements, end-to-end delay is an important parameter as it includes the total time required for multimedia processing, starting from capturing, digitizing, and encoding/compressing audio and video data, to transporting them from the source to the destination, and finally decoding and displaying them to the user. The conversational application requires low delay because it is a two-way conversation, while the rest of the two applications can tolerate the delay requirements [5][6].

The fundamental Internet infrastructure cannot adequately fulfill the Quality of Service (QoS) requirements of multimedia communications [7]. Recognizing this limitation, numerous research and development endeavors have been dedicated to establishing mechanisms that can ensure QoS requirements. The goal of this study is to present a thorough review of the numerous strategies and methods that have been created and used to guarantee QoS. The study examines technological developments made to address QoS issues in multimedia communications as well as strategies used to enhance the delivery of multimedia material over the Internet.

The rest of the paper is organized as follows: Section 2 provides an overview of different multimedia services; Section 3 delves into the discussion of metrics used to measure Quality of Service (QoS); Section 4 focuses on exploring the various approaches that guarantee QoS; and finally, Section 5 concludes by summarizing the key findings and implications.

2. Multimedia Services

Text, animations, graphics, audio, photos, and video are just a few examples of the various types of media that make up multimedia. Multimedia communications have been seen to significantly increase recently, with people creating this data as well as using it through the media files that are already available online [8]. To illustrate, in 2023, statistics showed that nearly 1 billion hours of content are watched across the world every day, and more than 500 hours of new content are uploaded to YouTube every minute [9]. Multimedia applications can be classified in different ways as shown in Figure 1.

Conversational applications: Applications such as voice services place a significant emphasis on preserving consistent timing among various data elements within the data stream. These applications exhibit a high sensitivity towards potential problems such as delays, jitter, and data loss. Ensuring minimal disruptions in timing and data integrity is critical for delivering a seamless and satisfactory user experience.

Streaming applications: Video streaming applications, although relatively more tolerant of data loss, remain sensitive to delays and timing variations, known as jitter. While they can handle some level of missing data, ensuring minimal delays and consistent timing is crucial to maintaining smooth playback and user satisfaction.

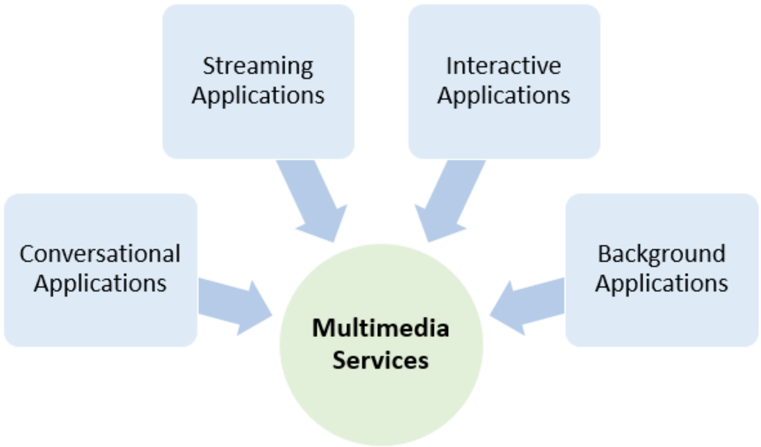

Figure 1. Classification of Multimedia Applications

Interactive applications: Web browsing applications function on a best-effort basis and typically follow a request-response model. Unlike conversational and streaming applications, web browsing is relatively less sensitive to delays. While delays may impact the user experience, the emphasis is placed more on delivering accurate and timely responses to user requests than real-time interactions.

Background applications: Email services, unlike time-critical applications, operate without the expectation of immediate delivery. Recipients of email services do not have specific time requirements for receiving messages. The emphasis is more on reliable message delivery than real-time transmission, allowing flexibility in when users can access and process their emails according to their convenience.

3. Quality of Service (QoS) Metrics

Quality of Service (QoS) is referred to as the overall effect of service performance that determines the degree of user satisfaction in ITU-T Recommendation E.800 [10]. QoS is the ability to provide for the requirements of various applications, data flows, and users by guaranteeing a specific level of performance for data traffic. QoS becomes essential in the context of multimedia applications for the uninterrupted transmission of multimedia content. When network capacity is constrained, it becomes especially crucial. Transmission guarantees are necessary for real-time multimedia transmission, including online games, IPTV, video conferencing, and Internet telephony.

Different applications may have varying requirements, such as minimal latency, reliable response time, or high image quality. The transmission of multimedia content, either in real-time or delayed, was significantly considered in several quality of service (QoS) metrics. These metrics encompass factors such as throughput, latency, jitter, packet loss rate, and bit-error rate [11]. These metrics are identified in Figure 2.

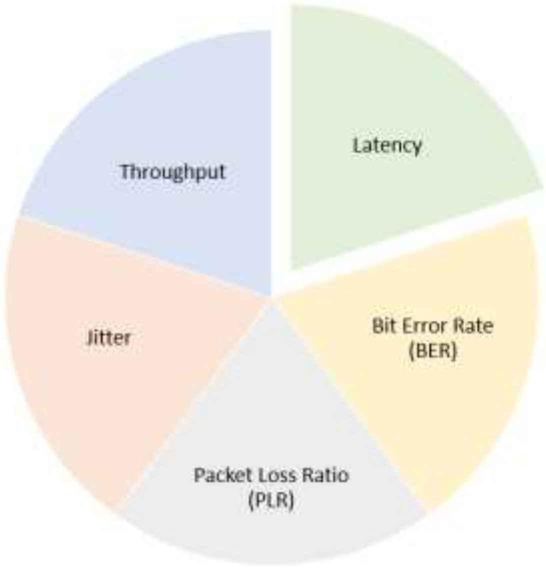

Figure 1. QoS Metrics for Multimedia Transmission

3.1 Throughput

Throughput is a key metric in assessing Quality of Service (QoS) that gauges the speed and efficiency of data transmission within a communication system. Higher throughput signifies a superior rate of successful data transfer, indicating faster and more efficient overall performance.

3.2 Latency

In various applications, end-to-end latency plays a crucial role as it affects end-user satisfaction. This latency is influenced by factors such as the speed of light and the functionality of intermediate network nodes like routers. High latency results in delays, causing an unfavorable user experience due to slower response times and decreased real-time interaction capabilities.

3.3 Jitter

In networks aiming to facilitate real-time audio and video, jitter emerges as a critical performance metric. It refers to the variation in delay experienced by packets during transmission. Real-time audio, in particular, is highly vulnerable to network jitter, primarily caused by inconsistencies in packet arrival times between the sender and client sides. As a result, the client receives packets with varying delays, leading to irregularities in the playback, synchronization issues, and disrupted audio quality, which collectively contribute to jitter.

3.4 Packet Loss Ratio (PLR)

The Packet Loss Ratio (PLR) is a crucial element that detrimentally affects the Quality of Service (QoS). Multimedia services, in particular, have a predetermined maximum acceptable threshold for loss. When the PLR exceeds this threshold, it leads to missing or corrupted data packets, resulting in degraded audio and video quality, disrupted playback, and an overall diminished user experience.

3.5 Bit Error Rate (BER)

The term "bit error rate" (BER) is defined as the number of bit errors over a unit time period. It is calculated by dividing the total number of bit errors by the total number of bits transmitted during the specified time period and is typically expressed as a percentage. Because lower error rates signify more precise and reliable data transmission, BER is an essential metric in assessing the dependability and caliber of a communication system.

4. QoS Requirements for Multimedia Applications

The end user requirement for Quality of Service (QoS) in multimedia communications varies across different applications, including real-time audio and video, where factors like low latency, minimal jitter, high throughput, and low packet loss are crucial to ensuring a satisfactory user experience and seamless data transmission. Various multimedia communications are given in Table 1 [7].

Table 1. QoS Requirements for End User

Medium	**Application**	**Type**	**Key Performance Parameters and Target Values**		
			End-to-end one-way delay	**Delay variation within a cell**	**Information loss**
Audio	Conversation	Conversational	< 150ms	1ms	< 3%
Video	Video Call	Conversational	< 150ms	NA	< 1%
Data	Interactive Games	Interactive	<250ms	NA	Zero
Audio	Voice Message	Interactive	< 1s	< 1ms	< 3%
Data	Transaction Service	Interactive	< 4s	NA	Zero
Data	Email	Interactive	< 4s	NA	Zero

5. Quality of Service (QoS) Metrics

Data is supplied either live or recovered from storage devices in a straightforward multimedia communication scenario. Data is then packetized in the transport module and sent over the Internet. Using network addresses, each packet is separately routed from its source to its destination. The multimedia data is put back together and provided to the appropriate application when it reaches its destination.

Regarding Quality of Service (QoS) guarantees in this communication scenario, we can summarize the main approaches as follows: To guarantee QoS, we can increase the speed of the Internet by deploying newly the proposed optical wavelength-division multiplexing (WDM) and OFDM (orthogonal frequency division multiplexing) technologies [12-14]. Another method to improve QoS is deploying various communication architectures like IntServ, DiffServ, and Multiprotocol Label Switching (MPLS).

Thirdly, it is often necessary to transmit data from a single source to multiple destinations. To optimize bandwidth usage and improve QoS, efficient multicasting protocols are required. These protocols help in minimize the amount of bandwidth needed for transmitting data to multiple destinations simultaneously.

Lastly, it is crucial to eliminate delay jitter at the destination before playing the data. Delay jitter can arise due to various factors, including disparities in packet processing times, network access times, and queuing delays. To address this issue, a first-in-first-out (FIFO) buffer can be employed at the destination. This buffer helps in mitigating delay jitter by organizing the incoming packets in the order they were received, ensuring smoother playback of the data.

In practical implementation, it is essential to incorporate all of the aforementioned approaches to effectively and efficiently achieve Quality of Service (QoS) guarantees. By implementing these approaches collectively, the desired level of QoS can be attained, ensuring optimal performance and user satisfaction.

In the subsequent sections, all the approaches mentioned above to guarantee QoS have been discussed in detail.

5.1 IntServ QoS

The underlying principle of the Integrated Services (IntServ) model is to allocate and reserve resources, such as bandwidth and memory, to fulfill the Quality of Service (QoS) requirements of an application or communication session, resulting in improved performance and user experience [15].

An application defines its Quality of Service (QoS) needs and sends them to the system using the IntServ model. The system will assess whether it has the resources necessary to meet these criteria. If the system has enough resources, it will accept the application and assign the resources required to guarantee that the QoS requirements of the application are satisfied. However, the system has two choices if it lacks the resources. It can either flatly refuse the application or suggest a more flexible QoS requirement. In the latter scenario, the application is approved and executed with a lower QoS level if it accepts the altered QoS settings. The application is denied and may not be accepted if it rejects the suggested parameters. The main advantage of the IntServ model is that it guarantees the Quality of Service (QoS) requirements for a session, provided the required resources have been successfully reserved. However, ensuring QoS without depending simply on allocating resources based on peak-bit rates is a significant difficulty. The objective is to provide QoS while effectively sharing and using resources, encouraging efficient resource use. The IntServ paradigm cannot be adopted gradually because all routers must set aside adequate resources to provide constant end-to-end Quality of Service (QoS), which is another problem.

5.2 DiffServ QoS

By classifying packets into a predetermined number of groups, Differentiated Services (DiffServ) establish a compromise between the IntServ service and the best-effort service. These classes are used to categorize all traffic packets, and each packet class has specific services provided by routers [16].

The differentiated service (DS) field replaces the type-of-service byte in DiffServ. The DS Code Point, which is the first six bits of the DS field, defines the desired behavior for each router's unique packets. The best-effort service is still provided to packets with a DS Code Point of zero. A slow rollout of DiffServ into the current Internet infrastructure is made possible by values between one and seven, which are created to ensure compatibility with the old IP precedence scheme. The client or transmitter process might designate the DS field to represent the desired service for the packets.

DiffServ ensures that 10 Mbps expedited traffic receives reserved resources, allowing it to traverse the network with minimal delay at any time. Traffic requiring assured service is prioritized and served before best-effort traffic, ensuring a specified level of QoS. The network allocates a bandwidth of 10-40 Mbps to serve the best-effort traffic, which is not time-sensitive or cost-effective to transport. This approach satisfies customers' needs and results in improved overall service quality for all customers.

5.3 Multiprotocol Label Switching (MPLS)

In an IP network, routers analyze the packet's header and use a routing algorithm to determine the next hop for the packet. This process involves two steps: classifying packets into forwarding equivalence classes (FECs) and mapping each FEC to a next hop. In Multiprotocol Label Switching (MPLS), the assignment of an FEC is done once at the ingress router and encoded as a label, which is inserted into the packet. Subsequent routers use the label as an index to determine the next hop and a new label.

Packets can be sent without every hop analyzing the network layer header thanks to this label-switching mechanism. A series of labels determines the path taken by the packet, known as an LSP (label-switched path). Label-switching routers (LSRs) are routers that are capable of supporting MPLS. The provision of differentiated QoS for various types of traffic is made possible by MPLS, which provides quicker forwarding than IP routing and permits differentiated handling of packets based on their labels.

MPLS combines aspects of both IntServ and DiffServ. It allows for the negotiation and reservation of a dedicated path to provide QoS guarantees, similar to IntServ. Additionally, it enables the aggregation of multiple flows under the same label, improving scalability, similar to DiffServ.

5.4 End Systems

The term “end systems” refers to the hosts of clients and servers in a network. These end systems consist of two fundamental components: a hardware architecture and an operating system. The hardware design encompasses processor power, memory capacity, data transfer bandwidth, and other resources. On the other hand, the operating system is necessary to efficiently manage these resources.

The operating system manages operations like resource allocation, scheduling, and prioritization, enabling effective use of the resources available. The operating system can minimize lags, enhance response times, and enhance system performance by skillfully managing the hardware resources.

The hardware architecture and the operating system work hand in hand to provide a robust foundation for QoS. Through proper coordination, they enable smooth data transmission, enhance network reliability, and support the various QoS parameters required by different applications. A well-designed and well-managed end system contributes significantly to delivering a satisfying user experience and ensuring efficient utilization of network resources.

6. Conclusions

In conclusion, the market for multimedia material is expanding quickly, and people are using the internet more frequently for communication and pleasure. This has brought to light the crucial role that Quality of Service (QoS) plays in delivering positive multimedia experiences. QoS requirements for multimedia services encompass flexible bandwidth allocation, minimized packet travel time, reduced jitter, and the ability to tolerate a certain level of data loss. These parameters are interrelated and vary based on the type of multimedia application. While the fundamental Internet infrastructure falls short of meeting QoS requirements, extensive research and development efforts have been dedicated to addressing this challenge. Various approaches and techniques have been discussed to ensure QoS guarantees and improve the delivery of multimedia content over the Internet.

References

[1] L. C. Wolf, C. Griwodz, and R. Steinmetz, "*Multimedia communication*", Proceedings of the IEEE, vol. 85, no. 12, December 1997, pp. 1915-1933, doi: 10.1109/5.650175.

[2] H. Sinnreich and A. B. Johnston, "*Internet communications using SIP: Delivering VoIP and multimedia services with Session Initiation Protocol*", John Wiley & Sons: New York, USA, July 2012, ISBN: 978-1-118-42915-0.

[3] T. Zahariadis, G. Pau, and G. Camarilo, "*Future media internet*", IEEE Communications Magazine, vol. 49, no. 3, March 2011, pp. 110-111, doi: 10.1109/MCOM.2011.5723807.

[4] R. A. Cacheda, D. C. García, A. Cuevas, F. J. G. Castano, J. H. Sánchez, G. Koltsidas, V. Mancuso, J. I. M. Novella, S. Oh, and A. Pantò, "*QoS requirements for multimedia services*", in *Resource Management in Satellite Networks*, G. Giambene, (Eds), Springer: Boston, MA, 2007, pp. 67-94, doi: 10.1007/978-0-387-53991-1_3.

[5] D. Kanellopoulos, S. Kotsiantis, and P. Pintelas, "*Internet and multimedia communications*", in *Encyclopedia of Information Science and Technology*, Second Edition, M. Khosrow-Pour (Eds), IGI Global, 2009, pp. 2176-2182. IGI Global: Pennsylvania, USA, 2009, doi: 10.4018/978-1-60566-026-4.ch343.

[6] N. Abbas and F. Yu, "*A comprehensive analysis of the end-to-end delay for wireless multimedia sensor networks*", Journal of Electrical Engineering & Technology, vol. 13, no. 6, November 2018, pp. 2456-2467, doi: 10.5370/JEET.2018.13.6.2456.

[7] Y. Chen, T. R. Farley, and N. Ye, "*QoS requirements of network applications on the Internet*", Information Knowledge Systems Management, vol. 4, no. 1, January 2004, pp. 55-76, doi: 10.1145/1030194.1015475.

[8] L. Skorin-Kapov, M. Varela, T. Hoßfeld, and K. T. Chen, "*A survey of emerging concepts and challenges for QoE management of multimedia services*", ACM Transactions on Multimedia Computing, Communications, and Applications (TOMM), vol. 14, no. 2s, May 2018, pp. 1-29, doi: 10.1145/3176648.

[9] Statistica, "*YouTube: hours of video uploaded every minute 2022*", www.statista.com/statistics/259477/hours-of-video-uploaded-to-youtube-every-minute/ (Accessed March 1, 2023).

[10] K. Koniuch, S. Baraković, J. B. Husić, K. De Moor, L. Janowski, and M. Wierzchoń, "*Top-down and bottom-up approaches to video Quality of Experience studies; overview and proposal of a new model*", Multimedia, arXiv preprint, January 2023, doi: 10.48550/arXiv.2301.11648.

[11] S. Gorlatch and T. Humernbrum, "*Enabling high-level QoS metrics for interactive online applications using SDN*", in 2015 International Conference on Computing, Networking and Communications (ICNC), Garden Grove, CA, USA, February 16-19, 2015, pp. 707-711, doi: 10.1109/ICCNC.2015.7069432.

[12] Y. Wang, Y. Wang, and N. Chi, "*Experimental verification of performance improvement for a gigabit wavelength division multiplexing visible light communication system utilizing asymmetrically clipped optical orthogonal frequency division multiplexing*", Photonics Research, vol. 2, no. 5, 2014, pp. 138-142, doi: 10.1364/PRJ.2.000138.

[13] Keysight Technologies, "*Concepts of Orthogonal Frequency Division Multiplexing (OFDM) and 802.11 WLAN*", www.rfmw.em.keysight.com/wireless/helpfiles/89600b/webhelp/subsystems/wlan-ofdm/content/ofdm_basicprinciplesoverview.htm (Accessed February 6, 2023).

[14] L. Lundberg, M. Karlsson, A. Lorences-Riesgo, M. Mazur, V. Torres-Company, J. Schröder, and P. A. Andrekson, "*Frequency comb-based WDM transmission systems enabling joint signal processing*", Applied Sciences, vol. 8, no. 5, May 2018, pp. 718, doi: 10.3390/app8050718.

[15] A. O. Adebayo and B. Twala, "*QoS functionality in software defined network*", in 2017 International Conference on Information and Communication Technology Convergence (ICTC), Jeju, Korea (South), October 18-20, 2017, pp. 693-699, doi: 10.1109/ICTC.2017.8191068.

[16] J. Zhao, N. Feng, and D. Ren, "*Research on energy efficiency based on differentiated QoS in fiber-wireless broadband access network*", Optik, vol. 126, no. 3, February 2015, pp. 350-355, doi: 10.1016/j.ijleo.2014.09.001.