QOS of Video Transmission

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Abstract: - This project demonstrates an approach to distributing video-on-demand (VOD) across multiple Video-on-Demand (VOD) instance servers using a client-server streaming technique based on the machine learning decision tree algorithm. One main challenge with using this technique is that a client-server approach must both efficiently utilize the strength of video data from the server(s) being requested at the same time as providing a means for asynchronous communication between the client-server video-on-demand (VOD) streaming instance servers and the video data being streamed from the server(s). In addressing this issue, we are able to provide a viable solution for robust recovery via parallel VOD video streaming instance server filling techniques, and encourage clients to maintain required levels of connectivity through alternative server score calculations. We have developed analytical models and classified server(s) according to different server type group classifications with respect to the capabilities the server(s) have available. By allowing QoS services to be utilized as a mechanism to alter traffic flows to avoid throttling capabilities of critical applications when the available network resources are limited. By using QoS technology, organizations can shift their overall network usage by providing preferential priority for specific, high throughput or high performing applications. Implementing parallel VoD streaming into parallel VOD instance servers that contain different buffer zones for providing incentive to maintain high levels of client peer connectivity will also provide insight concerning the relationship among efficiency, fairness, and incentive within the VOD client-server video-on-demand (VOD) streaming based upon the provisioning of QoS services. Through analytic analysis we demonstrate that robust Quality of Service (QoS) can be attained between clients and servers. The proposed method models the streaming process as an aggregation of end-to-end latency, bandwidth utilization, buffer occupancy, and packet delivery behavior analysis. The streaming rate is dynamically adjusted in response to current real-time network conditions to avoid both buffer starvation and excessive re-buffering.

Key-Words: - Video-on-Demand (VoD), Quality of Service (QoS), Decision Tree, Quality of Experience (QoE), Adaptive Streaming, TCP/IP Networks, Buffer Management, Multimedia Communication, Network Congestion Control, Streaming Optimization network congestion, packet loss, and variable.

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1 Introduction

There has been a significant rise in demand for multimedia applications (e.g., video-on-demand, live streaming, online exams, and real-time entertainment) due to the rapid development of the Internet and a sudden increase in the availability of broadband networks [1]. As a result, video traffic accounts for a substantial portion of current network traffic, making efficient and effective transmission an increasing necessity in modern communication systems. Even with advances in new network technologies, video streaming remains highly susceptible to network conditions. In particular, variables such as fluctuating bandwidth, transmission delay significantly affect video streaming quality. Unlike with conventional data applications, video streaming is sensitive to latency and depends on an uninterrupted stream of data for smooth playback. Even small changes in either latency or bandwidth will result in buffering, interruptions of playback, and an unsatisfactory user experience or, most importantly, packet loss. Quality of Service (QoS) plays an

important role in multimedia communications networks and provides a means of ensuring the performance requirements of low end-to-end latency, large data rates, low packet loss, and reducing the occurrence of re-buffering.Various QoS service models are illustrated in Figure 1.

Conventional media streaming methods typically use a constant streaming rate established through fixed buffering techniques.

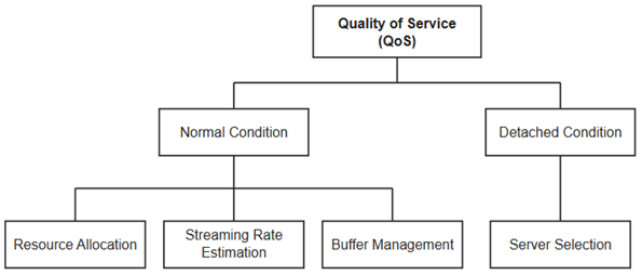


Fig. 1: QoS management under normal and detected conditions
Source: Various QoS service models

These methods mentioned cannot perform effectively in a range of networks as a result of their frequent changes between congested and non-congested network states and varying amounts of traffic. Therefore, the study proposes a video streaming application [2], [3], [4] that adapts to network conditions based on a quality of service (QoS) aware mechanism. The application will include both a delay-sensitive streaming rate control mechanism and buffer management mechanisms to ensure that playback is smooth.

2 Literature Review

With the explosive growth of Internet usage and the increasing popularity of various multimedia application types such as Video-on-Demand (VoD), streaming live video, cloud-based multimedia services, including eLearning platforms, these multimedia applications have created huge amounts of pressure on today's communication networks. Currently, video traffic accounts for a significant portion of total Internet traffic worldwide. Video requires a lot of bandwidth, very low latency, extremely little packet loss, and uninterrupted continuous playback.

Even with the continuing improvements or advancements in networking technologies, providing consistent Quality-of-Service (QoS) for video streaming customers is extremely difficult because of:

- Network congestion created by having too many users concurrently using the network, and large size video files.
- Network conditions that are highly dynamic and heterogeneous i.e. wired, wireless or mobile networks.

The proposed method would obtain real-time network conditions and dynamically adjust all video transmission parameters according to current network performance. By adapting streaming rates, bandwidth utilization, and buffer management in real time, the proposed system eliminates the need for large end-to-end buffers and reduces frequent buffering occurrences. This adaptive approach improves playback continuity, minimizes packet loss, reduces transmission delay, and enhances overall Quality of Service (QoS) during multimedia streaming.

In addition, the research tested the performance of the proposed application by collecting data on end-to-end delay, throughput, packet delivery ratios and buffering delays [5]. The results of the study showed that the adaptable nature of the proposed application improves the quality of streaming compared to normal streaming approaches by enhancing the quality of service (QoS), improving user friendliness, improves any future video streaming applications/ processes. Buffer underflow and overflow conditions, causing playback stalling.Server overloads and failures, especially in multi-server architectures;User mobility and intermittent connectivity resulting in degraded quality of streaming video. Typically, a traditional client/server video streaming architecture and current QoS-enhancing techniques focus on just one component, i.e. resource allocation, rate adaptation, or buffer management. As a result, the isolated techniques are unable to provide optimal performance when network conditions change dynamically, or when there are server failures.

3 Issue Definition

- Network resources that are allocated statically will create inefficiencies in the allocation of these network resources, as real-time channel conditions are not taken into account when allocating the network resources. This leads to resource waste, congestion and added time

delays.

- Existing rate adaptation methods do not provide sufficient adjustment to fast-changing bandwidth, leading to packet loss, buffering delays, and a degraded overall user experience.
- Buffers that are poorly sized and the lack of prioritization for packets both create interruptions in the playback of content, with an increase in latency, while at the same time decreasing the actual delivery ratio of content, particularly during peaks in the levels of congestion.
- The existing methods for selecting the optimum server for each user (in particular in multi-user scenarios) do not work well when servers fail or when users switch from one server to another, resulting in disruption of service to the users.
- Many architectures do not perform well in situations where users experience network disconnection, hand-off, or server crash, resulting in a significant loss of Quality of Service (QoS)/Quality of Experience (QoE). Work Related: This research aims to improve the Quality of Service (QoS) between a client and server during normal operation and disconnection conditions based on the following three parameters: delay, throughput, and packet loss. Effective resource allocation is essential to assure good Quality of Service (QoS) in video streaming applications.

With the availability of a multitude of streaming servers for streaming video, it is critical for clients to have options from which to select an appropriate server for the successful streaming of video. Although numerous streaming server options exist with copies of the same streaming video, the server that produces the best quality stream for clients will typically be the server that is in closest proximity to the client. The work of various authors has proposed various methods for identifying the best servers for use with various streaming applications. The protocol proposed by Bello, as referenced in other studies, uses logical mapping to create a distributed multimedia synchronisation protocol that synchronises logically and temporally interdependent data in real time [6], [7], [8]. The protocol described by Bello contains two primary components: the first is the computation of message deadlines using relative time points only, and the second is the partitioning of the processing stage of delivering multimedia in order to achieve synchronization based on an asymmetric design concept [9], [10]. The provision of their protocols, have resolved two main problems that have previously prevented mobile networks that utilize continually mobile devices from connecting to their networks. The first of these two problems is the number of times mobile devices get disconnected from their respective mobile network; as a result of the developer's work, it is predicted that the number of times that mobile devices get disconnected from their respective mobile network will be reduced significantly and in doing so, the overall Quality of Service of the mobile network will be greatly improved, and the cloud will be able to stream multimedia content to mobile devices without any interruption in a seamless manner. According to [11] they have addressed two primary limitations to the transmission of information by providing solutions through conditional integration with respect to saturation constraints encountered when transmission bandwidth/bits are available. The authors also identified the use of several heuristics for the design of a stable system, which will assist in alleviating Quality of Experience (QoE) issues. [12] indicated that they had developed a procedure for evaluating all the available paths of information flowing through their network from point of origin to destination using gray relational analysis. By assisting in determining the information quality along our network's different paths facilitates proper prediction of which path exhibits the highest potential for error as well as loss of quality of the video content being transmitted. Additionally, this information will assist the server in determining the appropriate path to use to transmit data. Experimental verification of this technique has demonstrated improved performance over several types of network channels. [13] designed and constructed a QOS architecture to address the maintenance of QOS requirements for mobile video surveillance systems using a 4G packet core that allows for faster transmission of data between media servers, QOS enablers, and machine to machine gateway systems. It is permissible for some systems to experience longer delays than current technologies while still preserving the required levels of quality for customers. Many real-time systems with a high success rate have had successful implementations. Research has also included the work of [14]. They proposed a Mobility Prediction Aware Bandwidth Reservation (MPBR); the purpose of the MPBR is to minimize the amount of new call barring while ensuring the minimum number of

hand-off calls. The MPBR consists of three components: 1) Hand-off Time Estimation (HTE), which provides the estimated hand-off time from time to time; 2) Available Bandwidth Estimation (ABE), which estimates the available bandwidth that the user will pass through until reaching his/her destination before that user makes his/her next hand-off; and 3) Efficient Call Admission Control (ECAC), which manages the distribution of bandwidth in the mobile network's cells. The evaluation of the MPBR system has been shown to provide better results than existing bandwidth assignment systems because there are fewer hand-off calls. In addition to this, allowing clients to choose the best servers in networked systems is critical to the achievement of high-quality service when there are network failures that create an unserviceable connection. There are many researchers who have addressed this issue, including Nodeja, 2001. However, the ability to select the best server for a particular user is found to be lacking in the above-discussed works.

4 System Overview

This section describes a streaming system in a client-server architecture, where multimedia content will be streamed from a video server over a TCP / IP network to one or more clients connected to the system. The Figure 2 shows multi server scenario for video transmission. Also included will be the integration and monitoring of the delays due to transmission time over the network, along with the management of the buffer management of packet data, and determining whether or not the packets arrive on time will be integral to the system.

The components of the proposed system are specified below:

- Video host.
- Video packet segmentation.
- TCP video data transmission.
- Video packet reception from the video server.
- Playback buffer.
- Monitor delay, packet arrival rate, and buffer state.
- Request a rate modification if conditions on the network deteriorate.

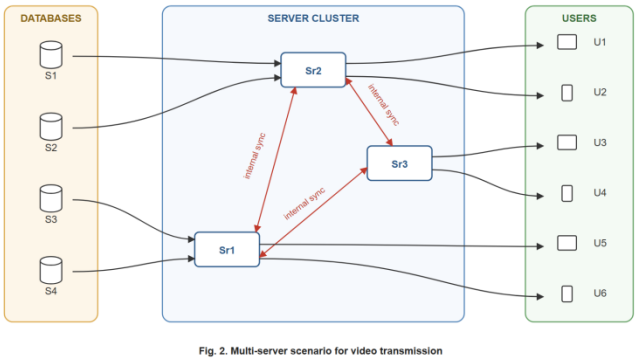


Fig. 2: Multi server Scenario
Source: [10]

5 Problem Formulation

The structure of this system is based on a client-server type of architecture. Video content from a video content server will be streamed to a client or multiple clients on a TCP/IP network. Furthermore, the proposed system will incorporate combining tasks related to monitoring delays and buffer management.

The system comprises:

- Video Hosting
- Video Segmentation into packets
- TCP transmission of Video Data
- Receiving Video Packets from Server
- Playback Buffer
- Delay, Packet Arrival Rate, and Buffer Occupancy Monitoring
- Requesting Rate Adaptation When Network Conditions Deteriorate

6 Problem Solution

The proposed methodology is as shown in Figure 3.
Step 1: Selecting Video and initializing Begin choosing and initializing video from the server side. A video will be selected to be streamed. It is broken down into pieces (or packets) to be transmitted through the IP network. Streaming parameters such as bit rate, packet size, and buffer threshold will also be set prior to beginning transmission.

Feature Representation

Let the input feature vector at time t be:

$$X_t = [D_t, B_t, T_t, L_t, Q_t]$$

Where

- D_t= End-to-end delay
- B_t= Available bandwidth

- T_t= Throughput
- L_t= Packet loss ratio
- Q_t= Buffer occupancy

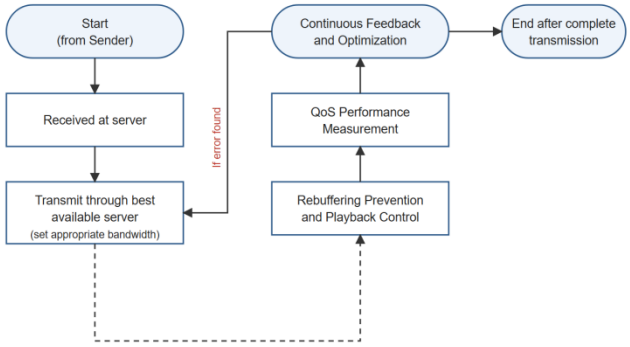


Fig. 3: Flowchart of major steps involved in the process
Source: Created by authors

- The target class label is:

$Y_t \in$ {Good,Moderate,Poor}

Step 2: The second step in video streaming involves sending video packets from a server to a client via the TCP protocol. The TCP protocol guarantees that all packets will arrive at their intended destination without errors and in the correct order, which is very important for smooth video streaming.

But when there is network congestion, TCP's ability to resend any lost (or dropped) packets adds additional delay due to the high rate of retransmissions, therefore requiring adaptive control from the application layer.

This streaming network is modeled as a graph as in Figure 4.

$$G = (N, L)$$

where: $L = \{l_1, l_2, \dots, l_m\}$ represents communication links and $N = \{n_1, _2, \dots, n_k\}$ represents streaming nodes (server, peers, clients) Each link l is characterized by:

- Bandwidth B_l
- Delay D_l
- Packet loss probability P_l

Monitoring of network conditions is accomplished through **continuous assessment.**

Step 3: As video packets move through the network, their path can experience many changes in the following conditions:

- Bandwidth availability
- Queuing delays
- Packet arrival rates

The client will continually monitor these conditions through observations of the arrival times of individual packets as well as packet throughput; using both of these methods of measurement, the client will calculate the total end-to-end delay and level of congestion within the network.

Step 4: Buffer Management at step 4 will take place at the client as well. The packets that were received will be buffered in a playback buffer in the client; buffer occupation will be continually updated based on:

- Rate of incoming packets
- The Rate of video playback

If the buffer level goes under a predefined minimum threshold, the client will classify this as having a risk of buffer starvation and interrupting video playback.

Step 5 will involve measuring and analyzing the end-to-end delay against a predetermined threshold. If the measured end-to-end delay exceeds the predetermined threshold, this indicates there is congestion on the network or insufficient bandwidth, which triggers the adaptive control mechanism.

Recursive Tree Construction

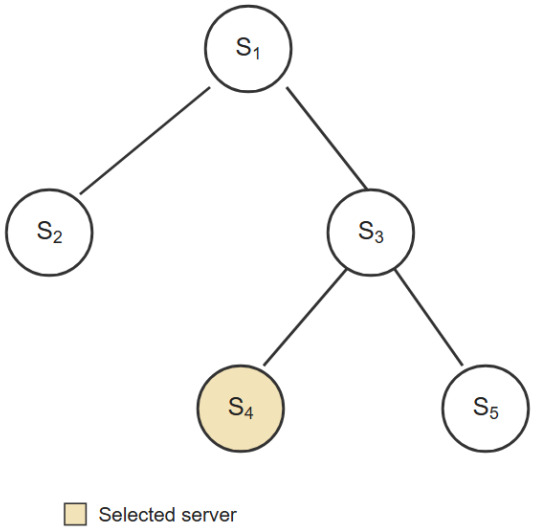


Fig. 4: Selection of server (Graphical representation of selection)
Source: [14]

The selection of server is as represented in Figure 4.

At each node:
1. Compute Gini for all features
2. Select the best feature for splitting
3. Partition the dataset
4. Repeat until a stopping condition is met

Stopping conditions include:
- Maximum tree depth reached
- Minimum number of samples per node
- Pure node achieved

Streaming Adaptation Logic

The predicted class guides streaming decisions:

$$Action = \begin{cases} \text{Increase or Maintain Bitrate,} & \text{if } \hat{Y} = \text{Good} \\ \text{Maintain Bitrate,} & \text{if } \hat{Y} = \text{Moderate} \\ \text{Reduce Bitrate and Increase Buffer,} & \text{if } \hat{Y} = \text{Poor} \end{cases}$$

Step 6: Adaptive streaming rate control: Based on current network conditions:
- Reduce streaming rate if latency and congestion are high.
- Increase streaming rate if latency and congestion improve.

The purpose of this dynamic adjustment is to find a balance between incoming data and available bandwidth. This will reduce packet loss and re-transmissions due to excessive delays.

Bandwidth Utilization Model

Let:= available bandwidth B_{req}= required video bit rate

Bandwidth ratio

$$BUR = \frac{B_{req}}{B_{avail}}$$

Streaming is considered **stable** if: ≤ 1

If $BUR > 1$, delay increases and video quality degrades.

Step 7: Re-buffering Prevention and Playback Control

Adaptive buffering strategies are applied to:
- Maintain minimum buffer levels
- Avoid frequent re-buffering
- Ensure continuous playback

The overall connection management is as shown in Figure 5.

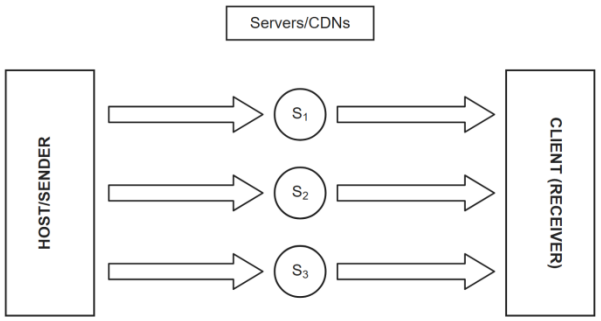


Fig. 5. Overall connection management

Fig. 5: Overall connection management
Source: Created by Authors

Total bandwidth equals sum of all server bandwidth

Playback is temporarily paused only if buffer starvation occurs, and it resumes automatically once sufficient data is buffered.

Information Gain

Information Gain determines the best feature for splitting the data.

$$IG(S,A) = H(S) - \sum_{v \in Value(A)} \frac{|S_v|}{|S|} H(S_v)$$

where:
- A= selected feature
- S_v= subset of data for value $v$of feature

The feature with **maximum information gain** is chosen as the decision node.

Step 8: Measuring QoS Metrics:Metrics associated with streaming are continually recorded as part of the QoS protocol:
- Throughput
- Buffering or download time
- Delivered Packets
- Total delay

Metrics will be used to evaluate performance of the system across different network conditions. Say D total is the total amount of time taken (end-to-end) for streaming a video by the formula:

where:
- D_{tx}=PacketSize /Bandwidth
- D_{prop}= propagation delay
- D_{queue}= queuing delay due to congestion
- D_{proc}= processing delay at intermediate nodes

Step 9: Continuous Feedback and Optimization
The client continuously sends feedback to the server regarding network conditions and buffer status. The server uses this feedback to optimize transmission behavior, ensuring stable streaming and improved Quality of Service.

Streaming Rate Adaptation Model:
To handle varying network conditions, streaming rate Rs is dynamically adjusted This adaptive behavior minimizes playback delay and packet loss. The Figure 6 and Figure 7 gives the influence of Bandwidth on Initial Play-out Delay and Figure 7 shows the influence of Server Failures on Failure Ratio.

The Decision Tree model relies on entropy, information gain, or Gini index to determine the optimal decision rule based on the parameters of QoS. Based on the classification of the network state as Good, Moderate, or Poor, the model allows the adaptation of the multimedia streams when they are buffered.

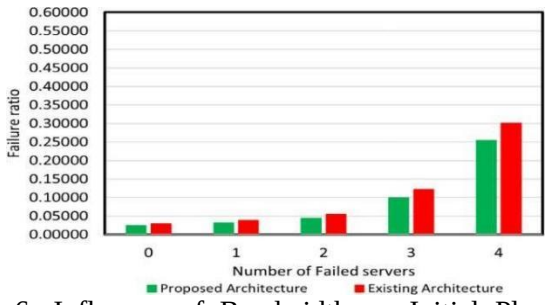


Fig. 6: Influence of Bandwidth on Initial Play-out Delay
Source: Created by Authors

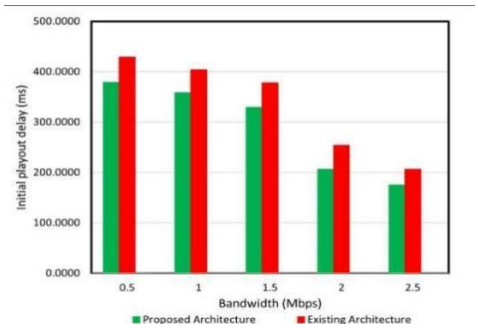


Fig. 7: Influence of Server Failures on Failure Ratio
Source: Created by authors

7 QoS Performance Metrics

- Throughput = Total Data Received/Total Time
- End to End delay = Summation D(tot) / Number of packets
- Buffering Delay = total Buffering Time / Total playback Time

8 Proposed Algorithm–Decision Tree

Step 1: The first part of the process is to gather and initialize training sets of network Quality of Service (QoS) attributes; these include end-to-end delay, bandwidth, throughput, packet loss ratio, and buffer usage/occupancy.
Step 2: The next step is to pre-process this gathered dataset; that is, you will handle any missing values, normalize/scale up/down all numeric variables (if required) and give a label (Good, Mod, Poor) based on how the network was experienced.
Step 3: Once the pre-processed data has been created, the actual development of a decision tree begins. The first step of developing the decision tree is to initialize the decision tree algorithm with the chosen criteria used to split nodes, for example, either (1) Information Gain or Gini Index, and (2) the termination criteria (i.e., Maximum Tree Depth and Minimum Number of observations per split). Once you have both impurities (Entropy & Gini Index) computed for the current dataset, you will then determine which attribute to use to split your node based upon the maximum Information Gain for each of the QoS attributes.
Step 4: During the fourth step of the development process you will compute both Entropy and Gini Index for the current dataset and will compute the Information Gain for each QoS attribute to determine which QoS attribute has the greatest Information Gain.
Step 5: Use the attribute that has the greatest Information Gain or the least Gini Index as your decision node, and then split the dataset using the thresholds associated with that attribute.

$$R_s(t+1) = \begin{cases} R_s(t) - \Delta R, & \text{if } D_{total} > D_{threshold} \\ R_s(t) + \Delta R, & \text{if } D_{total} < D_{threshold} \end{cases}$$

Step 6: The splitting process is then repeated recursively, recalculating impurity values for each sub-dataset, and choosing the optimal feature according to a stop criterion or a leaf node.

Step 7: Each leaf node gets classified into a Good,

Moderate, or Poor classification of Networks based on the majority of samples within that leaf node.
Step 8: As new parameters for Quality of Service in the live streaming video environment come into play, they are stored as the feature vector for classification.
Step 9: You will traverse from root to leaf as you look up the current prediction of the network based on the feature vector.
Step 10: The video stream rate / buffering mechanism can be adjusted to be consistent with the anticipated condition of the network.

For Good conditions:

- Bit rate.
- Increase/maintain bit rate for Good conditions
- Establish the bit rate for Moderate conditions
- Lower bit rates / longer buffering for Poor conditions

Step 11: Repeat the classification and adaptation process indefinitely until video streaming finishes or is stopped.

9 Dataset and Result

Our project was built using a single video, simulated to 1000 instances of server faults. A total of three TCP servers were used. Server optimization maximizes the client's ability to receive the finest video streaming service. Compared to the old design, there is a 6% diminish in end-to-end delay, 8% betterment in delivery ratio, and a 9% increase in throughput. The results are tabulated as shown in Table 1 which are comparative to P2P QoS Model.

Table 1. Comparative to P2P QoS Model.

Data Model	Dataset Quantity	End- end delay reduced in %	Throughput-increase	Overall betterment
Previous Model	To 1000 server faults	+0	+0	+0
Proposed Model	To 1000 server faults	+6	+9	+8

Source: created by authors

Table 1 illustrates the results are comparative to P2P QoS Model.

10 Conclusion

The number of people now accessing the internet and video streaming services has grown rapidly in the present day and age. Many video files are large, multiple video files are being transmitted at once, and they also require large amounts of bandwidth, which has added to congestion. To analyze and optimize the performance of video streaming services (Client-Server Architectures), both Server and client-based architectures can be optimised. The focus of the optimisation of the Server-side architecture will be on three modules: Resource Allocation, Streaming Rate Estimation, and Buffer Management. The Resource Allocation process will involve selecting a Queuing Discipline for accessing resources. The estimated congestion level, which has been determined by the use of HMM (Hidden Markov Model), is helpful in minimizing the loss of video packets that have been transmitted from the Source to the Destination (Client). To help reduce the number of packets being dropped during the Transmission of Video from the Source to the Client, the Streaming Rate Regulator Control Module (which utilises Fuzzy Logic Control) allows a different Transmission Rate to be assigned based on the amount of Congestion present in the Channel. Proactive Buffer Management will work to make Video Streaming Services more efficient and to reduce delays in Video Playback during periods of Congestion. Therefore, Quality of Service (QoS) is provided at both levels. By modifying the buffer size (and the rate) within the current architecture, Quality of Service parameters such as latency, throughput, and the Delivery Ratio of the proposed video streaming application architecture can be analyzed and evaluated. As such, efficient server optimization will maximize the amount of high-quality Video Streaming services that the client can receive from the stream. Results reflect a decrease in end-to-end delay by 7%, an increase in delivery ratio by 9%, and an increase in throughput by 11% when compared with the previous architecture. Further development can be made in relation to the optimization of the client side of the Server Section Module of the proposed architecture since external parameters affecting the QoS to be received by the client will be minimized as a result of the implementation of the proposed Server Section Control (SS-C) at the client side of the proposed design. Potentially applicable external parameters that can negatively impact our client's QoS are server failures or loss of Internet connectivity. The SS-C

will reduce the negative impact of server failures on QoS by calculating a fitness function and reducing the time of initial play-out and playback. SS-C demonstrated an initial play-out delay of less than 400 ms, which is an acceptable initial play-out delay for Video Streaming Applications, with minimum bandwidth and the maximum number of servers in detached mode. In summary, this method combines four distinct algorithms that are capable of achieving the highest possible quality video streaming experience for consumers. High Quality: All four algorithms have been created to optimize independent portions of the Video Streaming Architecture with the objective of providing the best video streaming overall. Thus, by having at least one algorithm optimized for each affecting factor of QoS (previously discussed factors) and having at least one algorithm that integrates the QoS-efficiencies of the independent algorithms, therefore creating a relatively high level of stability. The future research directions for this project will be focused on improving the Quality of Experience: QoE (a metric used to indicate the satisfaction level for customers consuming video streaming services). Factors that need consideration for developing an improved experience (QoE) focus on the interaction between: Humans, Systems, and Contexts. The ability of a video streaming service provider to manipulate the influence of their service system (to enhance the quality of experience) has to do with the provider affecting things directly related to the quality of video that is delivered and therefore needs to take into account the characteristics of the video being streamed (for example, the encoding and resolution) when it comes to invariant values.

Declaration of Generative AI and AI-assisted Technologies in the Writing Process

During the preparation of this work, the author(s) utilized artificial intelligence (AI) technologies solely for the purpose of linguistic refinement and grammatical correction. Following the application of these tools, the author(s) comprehensively reviewed, verified, and edited the resulting text. The author(s) maintain full accountability and responsibility for the accuracy, integrity, and final content of the publication.

References:

[1] K. Palanisamy, R. Venkatesan, M. Rajesh Babu, S. Uma, S. Rajeswari, and A. Kumar, AI Optimized QoS based On Demand Services with Incentive Mechanism in P2P Networks, Journal of Information Systems Engineering and Management,Vol. 10 No. 31s (2025), doi: 10.52783/jisem.v10i31s.5057.

[2] R. Venkatesan, Abdul Karim, Rajesh Babu, M., S. Uma, S. Rajeswari and Akash Kumar, "MPDM Scheme for QoS aware Routing protocol using Fuzzy Based Link Monitoring System", Journal of Information Systems Engineering and Management, Vol.10, No. 21s, 2025. https://doi.org/10.52783/jisem.v10i21s.3365.

[3] Devipriya K, Hemalatha R. (2023) Improving Quality of Serviceusing Multipath Routing Protocol for Delay Sensitive Applications of Internet of Things in Wireless Sensor Networks. Indian Journal of Science and Technology. 16(21):1538-1545. https://doi.org/10.17485/IJST/v16i21.441.

[4] Annapureddy, S., Guha, S., Gkantsidis, C., Gunawardena, D., and Rodriguez, P. (2007) Is high-quality VoD feasible using p2p swarming? In the 16th International World Wide Web Conference (WWW2007), 903-912. https://doi.org/10.1145/1242572.1242694.

[5] M.A. Bello, E.L. Dominguez, S.E. Hernandez, J. Cruz, Synchronization protocol for real time multimedia in mobile distributed systems. IEEE Access, 6, 15926-15940 (2018). https://doi.org/10.1109/ACCESS.2018.2817386.

[6] N. Al-shanableh, S. Anagreh, A.A. Haija, M. Alzyoud, M. Azzam, H.M. Maabreh, S.I.S. Al- Hawary, The Adoption of RegTech in Enhancing Tax Compliance: Evidence from Telecommunication Companies in Jordan. In Business Analytical Capabilities and Artificial Intelligence-enabled Analytics: Applications and Challenges in the Digital Era (pp. 181-195). Cham: Springer Nature Switzerland (2024).. https://doi.org/10.1007/978-3-031-57242-5_15.

[7] N. Fathima F. and H. Y. Vani, "Building, Deploying and Validating a Home Location Register (HLR) using Jenkins under the Docker and Container Environment," 2020 International Conference on Smart Electronics and Communication (ICOSEC),

Trichy, India, 2020, pp. 1278-1282, doi: 10.1109/ICOSEC49089.2020.9215458.

[8] A. Alzghoul, A.A. Khaddam, F. Abousweilem, H.J. Irtaimeh, Q. Alshaar, How business intelligence capability impacts decision-making speed, comprehensiveness, and firm performance. Information Development, 40, 220-233 (2024). https://doi.org/10.1177/02666669221108438.

[9] L. Shokr, D. AlAgry, S. Al-Sagga, Critical Assessment of Core Self-Evaluations Theory. Al-Balqa Journal for Research and Studies, 25, 162-184 (2022). https://doi.org/10.35875/1105-025-002-010.

[10] H.S. Lee, K.H. Lim, S.J. Kim, A configuration scheme for connectivity-aware mobile P2P networks for efficient mobile cloud-based video streaming services. Cluster computing, 16, 745- 756 (2013). DOI: 10.1007/s10586-013-0257-8.

[11] K. Miller, D. Bethanabhotla, G. Caire, A. Wolisz, A control theoretic approach to adaptive video streaming in dense wireless networks. IEEE Transactions on Multimedia, 17, 1309-1322 (2015). DOI: 10.1109/TMM.2015.2441002.

[12] M.M. Hassan, U. Farooq, Adaptive and ubiquitous video streaming over Wireless Mesh Networks. Journal of King Saud University-Computer and Information Sciences, 28, 432-446 (2016). DOI: 10.1016/j.jksuci.2015.11.005.

[13] M. Abu-Lebdeh, F. Belqasmi, R. Glitho, An architecture for `-enabled mobile video surveillance applications in a 4G EPC and M2M environment. IEEE Access,4, 4082-4093 (2016). DOI: 10.1109/ACCESS.2016.2592919.

[14] A. Nadembega, A. Hafid, T. Taleb, Mobility-prediction aware bandwidth reservation scheme for mobile networks. IEEE Transactions on Vehicular Technology, 64, 2561-2576 (2014). DOI: 10.1109/TVT.2014.2345255.

[15] J. Padhye, V. Firoiu, D. Towsley, and J. Kurose, "Modeling TCP Reno performance: a simple model and its empirical validation," *IEEE/ACM Transactions on Networking*, vol. 8, no. 2, pp. 133–145, Apr. 2000. DOI: 10.1109/90.842137.

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