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Visual Communication Design Method in Folk Art Based on Multimedia Data Transmission Technology

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ABSTRACT

With the rapid development of science and technology and China's economy, the internet, big data, computers, and multimedia technology are widely used in all walks of life to promote the application and improvement of visual communication design concepts, the gradual implementation of multimedia technology, and the continuous improvement of innovation. In this paper, from the multimedia transmission technology based on network coding, some key technologies in the network communication of data and video streams are studied in depth and, based on the transmission-quality assessment model, the working effect of the jitter-buffering algorithm and WebRTC jitter-buffering algorithm is compared in different network environments. The experimental results show that the jitter-buffering algorithm proposed in this paper has better working effect. This research is of great significance for the realization of multimedia transmission technology in next-generation networks.

KEYWORDS

Multimedia Technology, Visual Communication Design, Transmission Technology, Algorithm, Modeling

INTRODUCTION

The rapid development of multimedia technology has brought earth-shaking changes to the entire field of visual communication. This change requires us to adopt more innovative ways to meet user needs and drive social development. In this context, this paper aims to improve the quality of multimedia transmission and verify its practicality and feasibility by comparing the effectiveness of the proposed jitter-buffering algorithm with the WebRTC jitter-buffering algorithm in different network environments. To evaluate the transmission quality of different algorithms in network environments, we compared the proposed jitter-buffering algorithm with the original WebRTC system. The results show that in a normal network environment, the system transmission quality improved by 19.8% using the jitter-buffering algorithm proposed in this paper. In congested network environments, the transmission quality increased by 17.5%. This indicates that the jitter-buffering

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algorithm proposed in this article has better performance compared to the original WebRTC jitter-buffering algorithm in different network environments and can significantly improve network transmission quality.

These experimental results not only provide reference ideas and methods for subsequent research but also contribute to promoting the progress and development of multimedia technology. Through the jitter-buffering algorithm proposed in this article, we can effectively improve the quality of multimedia transmission and enhance the user experience. The research results of this article will provide valuable ideas and methods for subsequent research while also promoting the progress and development of multimedia technology.

LITERATURE REVIEW

Today, it is necessary to continuously implement the integrated development concept of visual communication technology and multimedia technology, meet the aesthetic standards and requirements of the public (Cecchinato et al., 2023). However, due to the constraints of traditional thinking modes, visual communication art designers are faced with heavy work pressure; the result is that the final design works displayed are at odds with the aesthetic art and thought concepts of current society (Silpa & Korra, 2023).

The application advantages of multimedia technology in the field of visual communication design have gradually emerged (Dhar et al., 2023). Using multimedia technology for related design can enrich the work. It has designed an intuitive and user-friendly user interface that allows users to easily interact with visual information, such as scaling, rotating, filtering, etc. For visual transmission based information visualization, consideration was given to how to transmit data in real-time and update visualization effects. (Jiang, 2023). The use of multimedia technology can not only enable designers to better showcase their design concepts, but also innovate the entire artistic design. (Gaikwad, 2024). Although multimedia technology has been greatly developed, it is still a major artistic medium for visual communication (Zhang & Cui, 2023). With the continuous development of this medium, multimedia technology and visual communication design can achieve deeper mutual promotion (Sharma et al., 2023).

All kinds of unprecedented electronic products have also changed the traditional ways of life of the public (Okpok & Kihei, 2023). For example, the emergence of robots, artificial intelligence, electric dishwashers, vacuums, etc. has improved people's standard of living but also made people more dependent on the conveniences conferred by technology (Mubarakali et al., 2023). New products are fun and addictive. However, it is also precisely because of the application of multimedia technology that more consumers and audiences have gradually increased their requirements for various new products with better use effects and wider functional coverage (Zhao et al., 2023). Existing high-tech products are constantly being upgraded. In view of the background of innovation and development, the visual communication design concept should highlight simplicity, and focus on the life of users. Satisfy user needs, so that consumers can feel the impulse to consume when they first see the product. (Yadav & Tiwari, 2023). Finally, visual communication designers should fully consider the use needs of consumers for the product and the possible expected uses when designing (Amiruzzaman & Bhuiyan, 2023). The design and style of the product can attract consumers and break through their psychological defense lines (Ahmed et al., 2023).

With the continuous improvement of China's production and development level, the functions of various electronic products are more abundant, the use effect is more convenient, and the design elements are more diversified (Makeeva et al., 2023). At the same time, the electronic products can also carry out personalized settings and function matching according to the actual needs of users (Mubarakali et al., 2023).

With more awareness of electronic products, consumers have more expectations of and thoughts about appearance and design. During the process of implementing the visual communication design concept of the product, designers should focus on the selection and design of functions and show the personalized development path of the product in a more ingenious way.

With the implementation, promotion, and popularization of the concepts of energy conservation, emission reduction, and green environmental protection in China, the application of various new energy sources is breaking through the constraints of the traditional production and development system (Benisha, 2023). Consumers pay more attention to environmental protection.

Second, when carrying out the innovative development of various product designs, we should optimize environmental performance, reduce energy consumption, improve energy utilization efficiency, and reduce secondary pollution.

Third, designers should not only think about the use requirements, selling points, and design of products, but also reflect the value and concept of environmental protection according to the innovative development direction of products and combine information technology to meet the requirements of environmental protection functions (Dhar et al., 2023).

Finally, when visual communication designers design and optimize products, they need to make consumers feel the ideas the designers are trying to convey and constantly feed their curiosity and desire to use products.

RELATED MATERIALS AND METHODS

Connotation and Characteristics of Multimedia Technology

Multimedia technology is based on the innovative development of advanced science and technology such as information technology and computer network technology (Taha & Ali, 2023). Multimedia technology has the characteristics of information carrier diversity (including text, pictures, animation, video, sound, etc.), information application integration, technical operation intelligence, information processing scalability, and information dissemination interactivity (Janakamma & Hegde, 2023). At present, with the continuous advancement of informatization construction, multimedia technology has gradually penetrated into people's daily life, production, and learning, providing technical support for modernization and sustainable competitive development in many fields such as games, art, entertainment, design, and education.

As far as visual communication design is concerned, the application value of multimedia technology is significant, which is embodied in the following aspects. First, it strengthens the information orientation of visual communication design (Kernen & True, 2023). That is to say, the application of multimedia technology in visual communication design can integrate and utilize a variety of information elements according to the requirements of visual communication design, give full play to the sensory stimulation of visual elements and auditory elements to information receivers, and give information receivers clear guidance.

Second, it improves the information-communication effect of visual communication design (Ahmed et al., 2023). In the process of traditional visual communication design, designers mostly use basic elements such as text, graphics, colors, symbols, etc. to present design ideas according to the requirements of visual communication design, combined with the characteristics of visual communication design objects. After receiving the information, the target group needs to understand the design intention through grasp the real information to be conveyed by the visual communication design. In this process, visual communication design is easily affected by factors such as the cultural background and ways of thinking of information recipients, which reduces the quality and efficiency of information transmission. The application of multimedia technology can make the design content be displayed in a more intuitive, vivid, and specific way, reduce the influence of the misunderstanding of information receivers on the effect of information transmission, and improve the quality and efficiency of information transmission.

Third, it enhances the artistic aesthetics of visual communication design and achieves visual balance (Mahmoodi Khaniabadi et al., 2023). Using multimedia technology in the process of visual communication design can reduce visual differences and improve visual balance by virtue of the characteristics of multimedia technology information digital dissemination, media integration, and utilization so as to cater to the thinking characteristics of different information receivers and meet their sensory experience needs. At the same time, with the scientific collocation of graphics, color, text, video, etc., the artistic aesthetics of visual communication design have been effectively improved.

Real-Time Multimedia Transmission Technology

Multimedia technology stores and transmits various media information such as sound, image, and animation and uses computer technology to process this information (Viola et al., 2023). In scenarios such as voice or video communication, the processing of sound and images has real-time requirements, which means the multimedia system must have good processing speed and efficiency. This paper summarizes the real-time multimedia transmission technology from the aspects of audio-compression technology, buffering technology, silence-detection technology, and video-compression technology.

Visual Communication Design

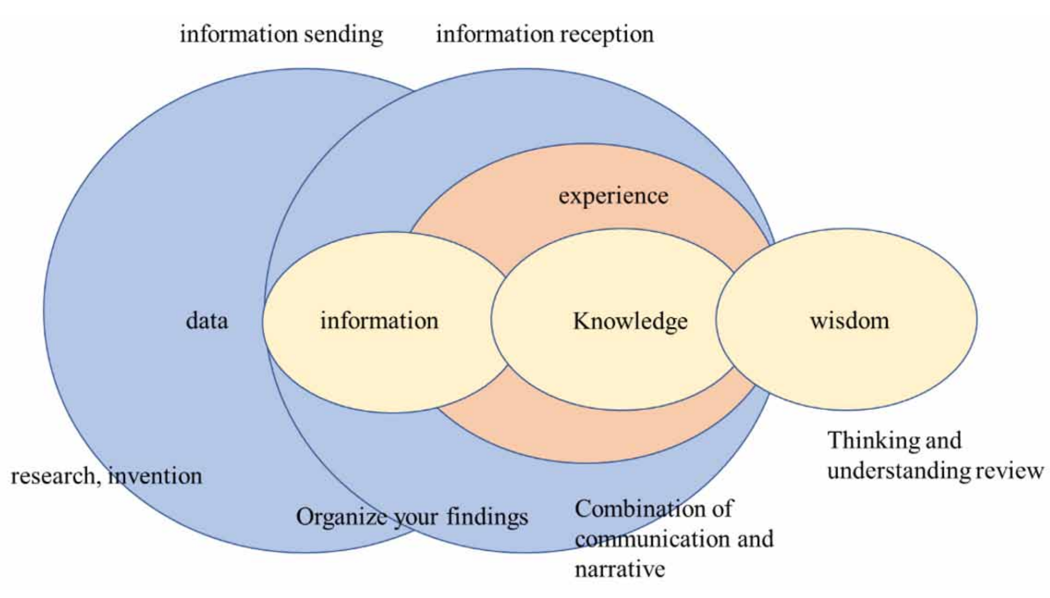
Communication refers to the process in which the sender of information uses symbols to transmit information to the receiver. Communication can occur within individuals or between individuals, such as between all living things, between humans and nature, between humans and the environment, and within the human body. It includes who, what to convey, to whom, effect, and how to influence these four procedures, as shown in Fig. 1.

Visual communication is interaction design, that is, communication design, which focuses on interactive experience, interactive feeling, and functionality.

From a professional standpoint, it mainly includes logo design, advertising design, packaging design, internal and external environment design, corporate image design, etc. Since these designs are communicated to consumers through visual images, they are called visual communication design. Visual communication design can vividly introduce a company's product information to consumers.

In this article, in order to collect and analyze data, we adopted the following specific methods:

Figure 1. The transformation process of information



(1) Data collection: we used WebRTC technology for video transmission and collected network parameters and video-quality evaluation indicators during the transmission process, including network latency, packet loss rate, video frame rate, etc. At the same time, we also conducted a survey questionnaire among the users participating in the experiment to understand their subjective experiences and feedback.
(2) Data analysis: we conducted statistical analysis on the collected data, compared the effects of different buffering strategies in different network environments, and used statistical methods such as stepwise regression and analysis of variance to determine the main factors affecting video transmission quality.

In the process of visual communication design, we have adopted the following specific techniques:

(1) Design thinking: in the design process, we adopted a design-thinking approach, which achieved rapid iteration and lean design through techniques such as deep understanding of user needs, exploring the essence of problems, and rapid prototyping design.
(2) User research: in the early stages of design, we conducted in-depth research and analysis on the target users to understand their needs, behaviors, and preferences in order to provide strong support for the design.
(3) Visual design: in the design process, we adopted visual design techniques, including color, layout, graphics, etc., to improve the readability and attractiveness of the design work while making it more in line with user needs and brand positioning.

In summary, we have achieved research and development of video transmission quality optimization algorithms through the use of data collection and analysis methods as well as design thinking, user research, and visualization design techniques in the visual communication design process. We have successfully applied them to practical scenarios.

Scalable Video Robust Transmission Method

Scalable video coding is a video-coding algorithm suitable for network transmission (Liu & Yao, 2023). Liu & Yao(2023) analyzed the performance in networks with packet loss. The generated composite hierarchical code stream includes a base layer and several enhancement layers. When the terminal receives the base layer and part of the enhancement layers, it can decode to obtain an image sequence with a certain amount of distortion. The hierarchical characteristics allow the scalable code stream to be segmented before transmission and then transmit the scalable code stream with better rate-distortion performance first, thereby improving the quality of the reconstructed image per unit of time. The existing code stream segmentation methods are divided into two categories: image frame-based segmentation methods and image group-based segmentation methods. The method based on image frame has better performance, but there are problems of high distortion calculation and high computational complexity, which makes it difficult to apply this kind of algorithm to the actual video system. The method based on codec takes the basic unit of encoding and decoding as the basic unit for processing the code stream, the algorithm is simple and easy to implement, and the code stream segmentation algorithm is based on the quality layer.

Since the division method of the scalable code stream in the spatial and temporal domains is not considered, it cannot be applied very well to the actual video system. Existing segmentation methods based on spatial domain, time domain, and quality domain do not consider the influence of transmission bandwidth on the ranking of scalable code streams. A global dynamic greedy algorithm is used to search for scalable code streams with the best rate distortion. When the bandwidth is low, the quality of the reconstructed video may be greatly reduced, so the layered code stream with the best average reconstructed video quality is selected, and its performance is not optimal under a given transmission bandwidth. Moreover, both methods use the reconstructed image sequence relative to the original image sequence to calculate the image distortion, which will also increase the computational overhead and delay of the system.

Support vector classifier (SVC) provides three scalable methods, namely, spatial domain grading, temporal domain grading, and quality domain grading. During source coding, the video sequence is divided into multiple groups, each group is called a group of pictures (GOP), and GOP is the basic unit of SVC coding. The composite hierarchical code stream generated by a GOP encoding contains a base layer and several enhancement layers. Each enhancement layer can be one of three enhancement layers in the spatial domain, time domain, or quality domain, and each enhancement layer can provide better resolution (spatial or temporal) or better quality reconstructed video sequences. Fig. 2 shows the principle of the SVC algorithm. As shown in the figure, the solid line path code stream reconstructs the video sequence with higher spatial resolution and lower temporal resolution.

Four times Common Intermediate Format (4CIF), Common Intermediate Format (CIF), and Quarter Common Intermediate Format (QCIF) are all standards for video image resolution and are commonly used in video encoding and transmission. JVT/AVC Software Reference Model (JSVM) is a software reference model for H.264/AVC video encoding standards. The data-segmentation algorithm provided by JSVM does not consider the rate-distortion performance of different reconstructed video sequences, so the algorithm cannot effectively improve the quality of reconstructed images. To improve the quality of the reconstructed image, when the sorting algorithm sorts the layered code stream, the rate-distortion characteristic of the layered code stream must be considered when the bandwidth of the terminal is R. The code stream sorting algorithm with the best rate-distortion performance can be expressed by the following formula:

$$\theta^* = \arg\min_{\theta \subseteq \Theta} D(\theta), \text{s.t.} R(\theta) \le R_B \tag{1}$$

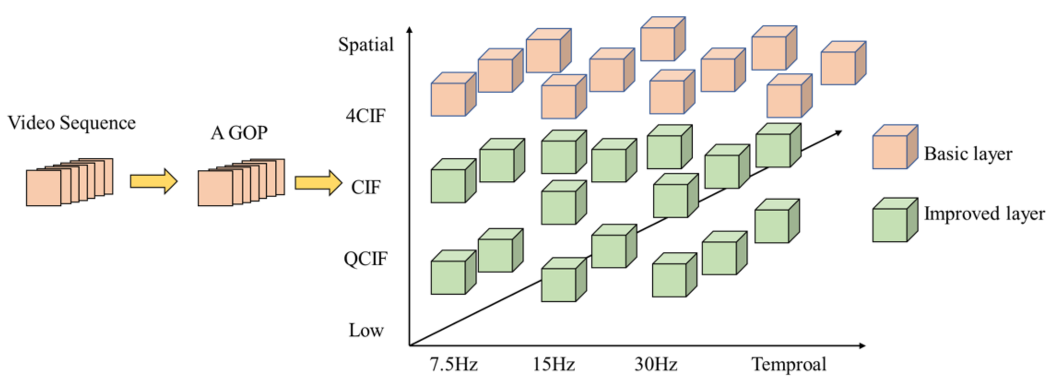
The SVC scalable flow state model established in this section satisfies the following properties:

1) $L(1 \le j < M)$, $Gk(1 \le k \le U)$, and $P(1 \le i < H)$ represent nodes, generations, and paths in the model, respectively, where N, U, and H are defined as

$$N = D_d \cdot T_t \cdot Q_q \tag{2}$$

$$U = D_d + T_t + Q_q - 2 \tag{3}$$

Figure 2. Schematic diagram of SVC algorithm



$$H = \frac{\left(D_d + T_t + Q_q - 3\right)}{\left(D_d - 1\right)!\left(T_t - 1\right)!\left(Q_q - 1\right)!} \tag{4}$$

L represents the layered code stream of SVC.

2) λ represents the set containing the worker and all its elder nodes, then λ satisfies:

$$\lambda_k^i = \begin{cases} \left\{L_1\right\} \\ \lambda_{k-1}^i \cup \pi_k^i \\ \left\{L_1, L_2, \cdots, L_N\right\} \end{cases} \tag{5}$$

Node (*di*;) represents the corresponding element in 5; *D* represents the element containing the distortion of the code stream.

3) Sets 12...21 are mapped to sets {9...re}; so that the distortion of λ in the path *P*(! ≤ *i* ≤ *H*) decreases as the generation increases, when *D*(*d, t, k*) > *D*(*d, t, q*).

Definition δ(1 ≤ *i* ≤ *H*, 1 ≤/≤ 0) represents the *i*th state in the path P_i in the state model. Its expression is

$$\delta_j^i = \begin{cases} \gamma_1^i \\ \gamma_j^i - \gamma_{j-1}^i \end{cases} \tag{6}$$

and the state δ satisfies

$$\begin{cases} \bigcup_{j=1}^{\theta_i} \delta_j^i = \bigcup_{\varepsilon=1}^{N} L_\varepsilon \\ \delta_n^i \bigcap_{n \neq m}^{1 \leq n,m \leq \theta_i} \delta_m^i = \varnothing \end{cases} \tag{7}$$

δ represents the scalable code stream contained in the *j*th state in P_i. The state model reflects the relationship between individual scalable code streams.

Robust Transmission Technology for Wireless Networks Based on Opportunistic Network Coding

With the development of wireless networks, especially mobile communications, more and more attention has been paid to wireless network transmission technologies (Li & Li, 2023). Among them, multicast and broadcast methods can enable multiple terminals to receive data sent by base stations at the same time. They are used as efficient data-transmission technologies. However, packet loss caused by multipath interference and fading of wireless channels seriously affects the efficiency of wireless broadcast transmission. Therefore, it is necessary to design a more effective retransmission method to improve the broadcast transmission efficiency of data in wireless networks. The existing retransmission methods can be divided into two categories: an automatic repeat request (ARQ) retransmission method and a network coding based ARQ transmission method, as shown in Fig. 3.

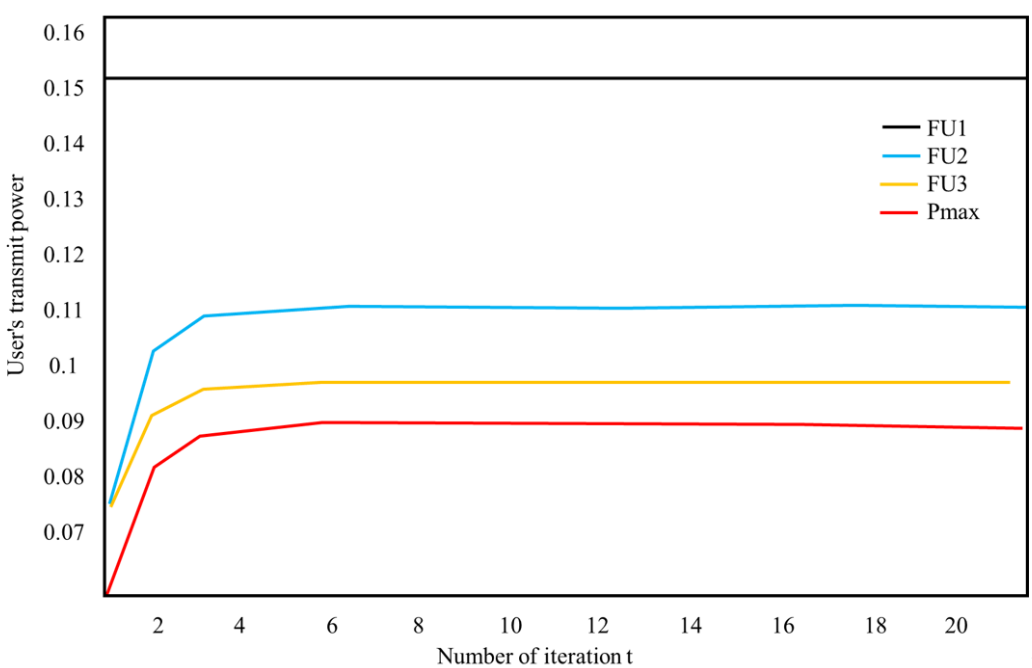
To improve the efficiency of wireless network data transmission and reduce the number of packet-loss retransmissions, this paper proposes a new broadcast retransmission method suitable for wireless networks. Using the packet-loss information fed back by the terminal, this method selects the lost packets at the base station based on the lower limit of the number of retransmissions and the maximum number of lost packets and then uses opportunistic network coding (ONC) to generate combined retransmission packets. The number of retransmissions is effectively reduced and a method of recovering lost packets from multiple retransmission packets is adopted at the terminal, thereby improving the efficiency of the terminal in recovering lost packets.

In the wireless network base station, the opportunistic coding selects multiple data packets for coding to generate a coding combination packet. The coding method is to XOR the bits of the selected data packets at the same position. Assuming that Xi $(1 < i < N)$ represents a data packet and Y represents a coding combination packet, Xi $(1 < i < N)$ and Y can be represented by binary sequences [$xi1$, ..., xij, ..., xiL], where L is the length of the sequence. Then, according to the encoding method, the Y sequence of the encoded combined data packet is generated, which can be described as:

$$y_j = \sum_{i=1}^{N} \mu_i x_{ij} \bmod 2, \quad \mu_i \in \{0,1\} \tag{8}$$

The wireless terminal can decode the packet loss from the received code combination packet. The decoding method is to XOR the received data packet and the bits of each position of the code combination packet to obtain the packet loss. It is worth mentioning that only when the encoding combination packet contains a packet loss of the terminal can the terminal decode the packet loss

Figure 3. Robust power allocation in uplink transmission femtocell heterogeneous network



from the encoding combination packet. Assuming *XI* represents a data packet, if the terminal receives *Y* and other data packets except for the dragon, the terminal obtains the *j*th bit x_j of the packet loss *XI* sequence, which can be described as:

$$x_j = \sum_{\beta \neq i} y_j \oplus x_{ij}, \ \ \beta = 1, 2, \cdots, N \tag{9}$$

In order to describe the transmission performance, the number of retransmissions *R*, the successful decoding rate *D*, and the throughput *T* are defined. *R* is represented as:

$$R \geq \max\left\{S_1\left(\Omega\right), \cdots, S_j\left(\Omega\right), \cdots, S_M\left(\Omega\right)\right\} \tag{10}$$

where *s* (Ω) represents the number of packets lost in *T*; *D* is defined as:

$$D = \frac{1}{M}\sum_{j=1}^{M} \delta_j\left(\varepsilon_j\right) \tag{11}$$

Among them, the contribution δ(ε) indicates whether the write has received all the data packets after ε_j retransmissions. If it is successfully received, its value is 1; otherwise it is 0. *T* is defined as

$$T = \frac{1}{M}\sum_{j=1}^{M} \frac{\varepsilon_j}{\gamma_j}, \ \ \varepsilon_j \neq 0 \tag{12}$$

where γ indicates that all data packets are received after γ_j retransmissions. *R* represents the number of packet retransmissions, and the smaller *R* is, the better the performance of the transmission algorithm. *D* represents the efficiency of data-packet transmission, and the closer *D* is to 1, the better the performance of the algorithm. *T* represents the efficiency of data-packet reception, and the larger the value, the better the performance of the algorithm.

When 0 < *C* < *M*, select packet loss for coding combination. Let *MX Yk*-dimensional matrix } *k* represent the information of *Yk* packets lost in the *k*th retransmission packet and its rows and columns represent *Yk* packets and the packet-loss information of the terminal, respectively. }*k* is generated by the following steps. The number of generated retransmission packets satisfies the constraint of the lower limit of the minimum number of retransmissions, and the lower limit of the minimum number of retransmissions corresponding to the packet loss information port is $Rfl = \alpha_1 + \alpha_2$. When generating the *k*th retransmission packet, the method in this section selects a certain number of lost packets so that when no packet loss is selected to generate a retransmission packet, the restriction condition under the minimum number of retransmissions is satisfied, that is:

$$S_j\left(W_r^*\right) = R_{0_1} - \alpha_1 - k \tag{13}$$

where *W** represents the matrix composed of unselected columns in Ω; *S*(*W**) represents the number of packets lost in the *j*th row of *W**.

$$S_j\left(\Theta_k\right) \geq 2,\ j \in \left\{1,2,\cdots,M\right\}_j \tag{14}$$

First, from the $\Theta_k(l' kca_2)$ generated in the second process, search for α3 data packets that the terminal cannot recover; α3 is

$$\alpha_3 = \sum_{j=1}^{M}\left(\lambda_j - 1\right), \lambda_j > 0 \tag{15}$$

where λ represents the number of unrecovered data packets written, and then α3 data packets are transmitted in sequence.

RESULTS AND ANALYSIS

Analysis of Experimental Results

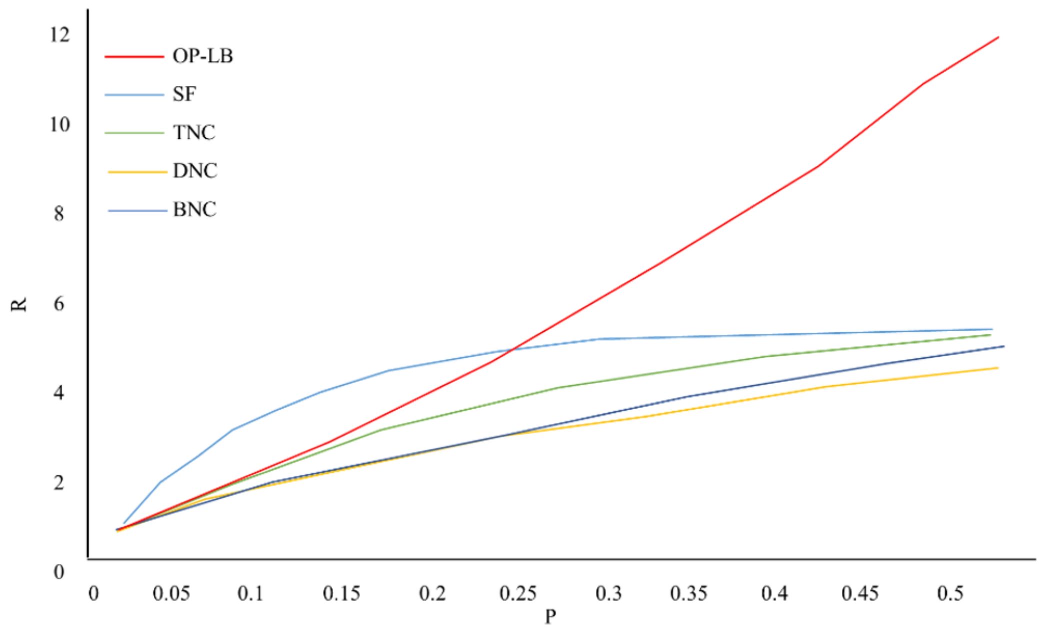
In the experiment, a multicast network is used, and the source node sends a retransmission packet to each terminal by broadcasting each time. The length of the retransmission packet and the time interval for sending the retransmission packet are the same. Under the condition of the same packet-loss information port, different broadcast retransmission methods generate retransmission packets according to their respective coding strategies, and the performance of different algorithms is obtained through simulation experiments.

The variation range of the packet-loss rate *P* is [0.02, 0.5], the step size is 0.02, the number of packets $N - -\}S, I0, 50\}$ the number of terminals *M* = 10, and the packet loss rate is *PJ* = *P*. Fig. 4 shows the effect of the change of the packet-loss rate on the number of retransmissions of the algorithm. With the increase in packet-loss rate, the performance of the broadcast retransmission algorithm based on ONC is obviously better than that of the traditional SF algorithm, the SF algorithm represents a dynamic grid resource scheduling algorithm that prioritizes meeting minimum service requirements. This algorithm prioritizes meeting the minimum resource requirements of existing tasks, thereby reducing the waiting time of individual tasks. Oncbr algorithm is an adaptive heuristic algorithm for solving combinatorial optimization problems with discrete constraints. The algorithm is based on backtracking and finds the optimal solution of the problem through continuous search and pruning. And the performance of the ONCBR algorithm proposed in this chapter is better than that of other algorithms. Under the condition of a certain packet-loss rate, the number of lost packets increases with the increase in the value of *N*. ONCBR chooses a better coding strategy for packet loss, so the average number of retransmissions *R* is the smallest, while the SF algorithm does not encode data packets.

In order to verify that the algorithm can effectively reduce the number of retransmissions and has better successful decoding rate and throughput performance, the random packet-loss model is used in the experiment. The packet-loss rate of different terminals is the same and is equal to the value of *q*. From 0.02 to 0.2, the step size is 0.02; whether the terminal loses packets or not is subject to the Bayern effort experiment with parameter *q*. The number of retransmissions and the successful decoding rate are shown in Fig. 5 and Fig. 6, respectively.

From the results in Fig. 5, it can be seen that the number of retransmissions of the proposed algorithm is always smaller than that of SF, TNC, and BNC. TNC: Refers to the fast retransmission mechanism in Transmission Control Protocol (TCP). Fast retransmission is a congestion control mechanism used to quickly recover lost data packets. BNC: This abbreviation is not a universal or widely recognized algorithm or technology. BNC interface refers to coaxial cable interface. DNC,

Figure 4. Comparison of retransmission times for the uncorrelated model of depacketization



also known as Distributed Numerical Control, is a commonly used manufacturing term for networked CNC machine tools. However, with the increase in *N*, the number of retransmissions is slightly larger than that of the DNC method. At the same time as the successful reception rate and throughput, the number of lost packets that cannot be recovered correctly by the terminal increases. With the increase in *N*, it increases slightly, so the number of retransmissions increases slightly. As shown in Fig. 6, with the increase in *N*, the successful reception rate and throughput of the proposed algorithm are always better than those of other algorithms. The reason is that this method selects more lost packets when generating the coded combination packet. On the one hand, it improves the transmission loss of the base station. On the other hand, the terminal recovers its lost packets from the coded combined packet with high efficiency, so the successful reception rate and throughput performance of this method are the best.

Analysis of Work Effectiveness

Based on the WebRTC multimedia real-time communication framework, a comparative experiment was designed to verify the above analysis. The experiment was divided into two sub-experiments by setting different network environments.

Comparative Experiment Under Normal Network Environment

The purpose of the experiment is to verify that the jitter-buffering algorithm based on machine learning proposed in this paper has a better working effect than the jitter-buffering algorithm in the original WebRTC.

Based on the transmission quality evaluation model proposed in this paper, the transmission quality is evaluated from three aspects: system packet-loss rate, end-to-end delay, and playback jitter; the jitter-buffering algorithm is controlled as the only variable. Quality is the feedback of the working effect of the jitter-buffering algorithm: if the system implementing the jitter-buffering algorithm proposed in this paper has lower packet-loss rate, end-to-end delay, and playback jitter,

Figure 5. Number of retransmissions

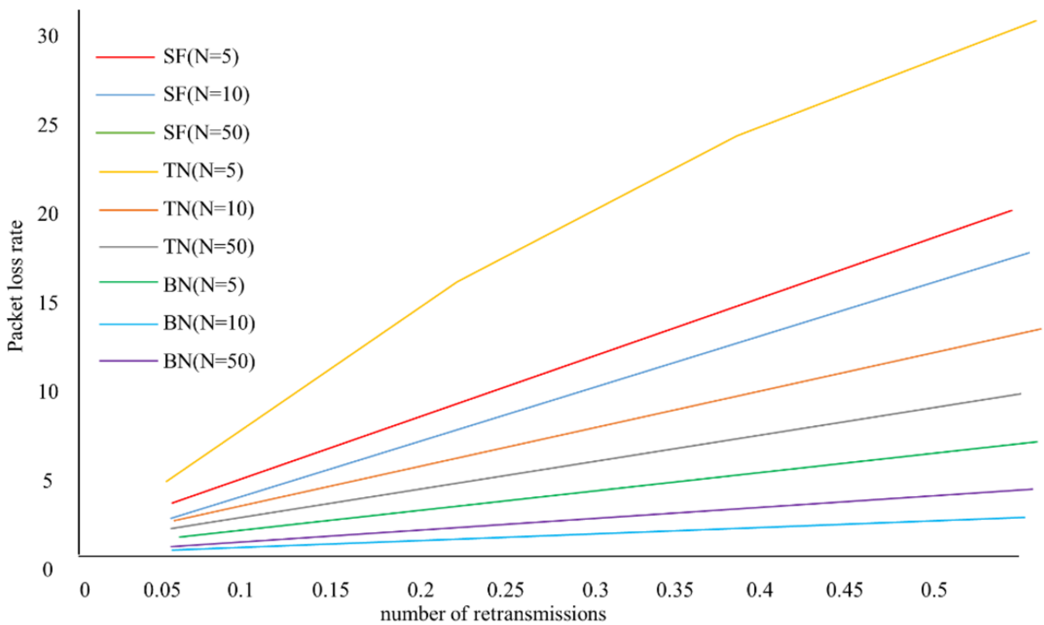
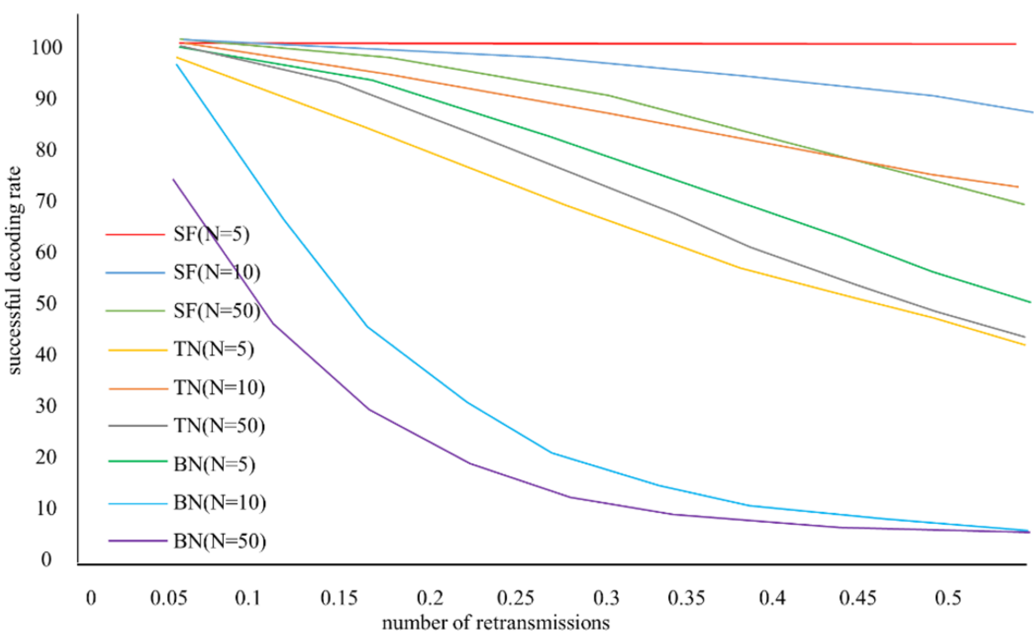


Figure 6. Successful reception rate



it shows that the system can provide higher transmission quality, thus verifying that compared with the original WebRTC jitter-buffering algorithm, the jitter-buffering module based on the machine-learning prediction model proposed in this paper has a better working effect.

Comparative Experiment in a Congested Network Environment

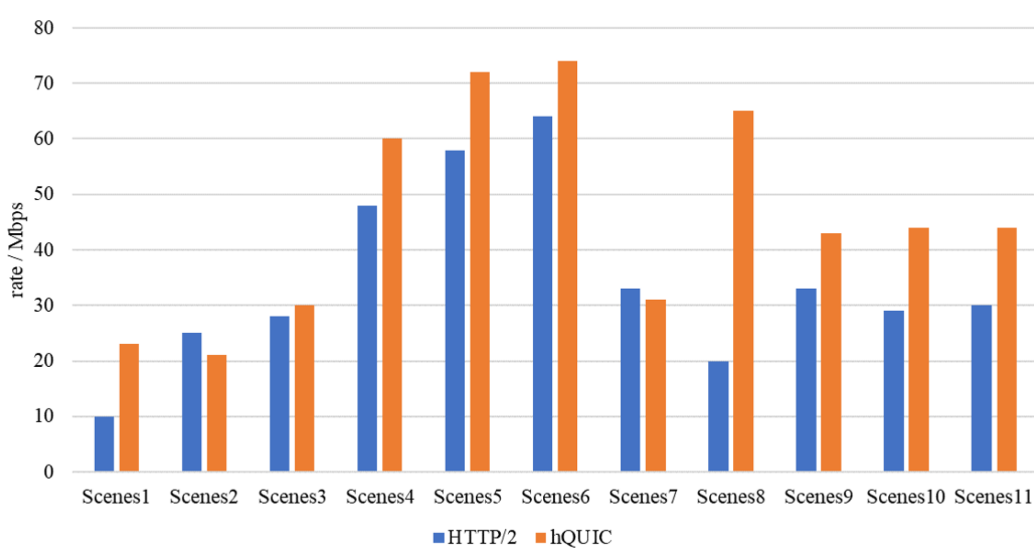
The purpose of the experiment is to verify that the machine learning based jitter-buffering algorithm proposed in this paper can be applied to different network environments. Based on the transmission-quality evaluation model proposed in this paper, the transmission quality is evaluated from three aspects, system packet-loss rate, end-to-end delay, and playback jitter, and used as feedback to verify the working effect of the jitter-buffering algorithm. We hope to show through comparative experiments that the systems that implement the jitter-buffering algorithm proposed in this paper have lower packet-loss rate, end-to-end delay, and playback jitter under different network environments. It shows that the jitter-buffering algorithm proposed in this paper updates the jitter delay according to the network conditions and has a better working effect than the original WebRTC jitter-buffering algorithm in different network environments.

Based on the research on the transmission-quality evaluation model in the analysis of experimental results, in this experiment, the following three quality-of-service indicators are selected to evaluate the experimental results: packet-loss rate, end-to-end delay, and playback jitter, which reflect the media quality, real-time performance, and stability in the system transmission process, respectively.

Before data processing, import six sets of log files into Jupiter notebook and use. Descripebe () to view the basic situation of these data. In order to correctly reflect the stability of the communication process, the absolute value of the playback jitter data is analyzed. The maximum and minimum values of packet-loss rate, end-to-end delay, and playback jitter in the six groups of data are recorded, and the experimental data under normal network environment and congested network environment are processed.

Figure 7 depicts the average packet-loss rate, end-to-end delay, playback jitter, and transmission quality of the WebRTC communication system using the jitter-buffering algorithm proposed in this paper and the original WebRTC communication system during the three experiments. Compared with the original algorithm, the packet-loss rate, end-to-end delay, playback jitter, and comprehensive improvement are 21.5%, 15.7%, 20.1%, and 19.8%, respectively; the optimization effect of this algorithm on transmission quality is obvious. Specifically, this algorithm can achieve these optimizations by processing and caching data during data transmission. When there is significant network delay or jitter, this algorithm can automatically adjust the data-transmission rate to adapt to the current network

Figure 7. Comparison of the results of the two algorithms in a normal network environment



conditions. Meanwhile, the algorithm can also correct errors in data transmission by using forward error correction techniques, thereby improving the reliability and stability of transmission. Overall, the jitter-buffering algorithm is a very valuable technology that can provide users with a more stable and high-quality transmission experience in WebRTC communication systems.

It is concluded from this experiment that under the same network environment, transmission media content, and codec used, the effect of the jitter-buffering algorithm will directly affect the transmission quality of the multimedia communication system, thus affecting the user's communication experience. Compared with the original WebRTC communication system, the WebRTC communication system using the jitter-buffering algorithm based on the machine-learning prediction model proposed in this paper has a lower packet-loss rate, lower end-to-end delay, and better performance in a normal network environment. Small playback jitter shows that the jitter-buffering algorithm based on the machine-learning prediction model proposed in this paper can obtain more appropriate jitter delay adjustment so as to provide higher quality, lower delay, and smoother video-communication services.

Using a recorded one-minute sample video as the video input source, three effective video communications were carried out, and the experimental results are shown in Fig. 8

It is concluded from this experiment that the jitter-buffering algorithm proposed in this paper is adjusted based on the current network environment. In different network environments, the system using the jitter-buffering algorithm proposed in this paper has lower loss than the original WebRTC communication system. Lower packet-rate loss, lower end-to-end delay, and smaller playback jitter can provide higher quality, lower latency, and smoother video-communication transmission. In different network environments, there may be differences in the stability and reliability of data transmission, such as network latency, bandwidth fluctuations, network congestion, and other issues that may affect transmission quality. The jitter-buffering algorithm can adaptively adjust the data-transmission rate according to the current network conditions to adapt to different network environments. This means that regardless of the network environment in which users use the WebRTC communication system, they can obtain a stable, efficient, and high-quality communication experience. In addition, jitter-buffering algorithms can also optimize the utilization of network bandwidth, reduce network congestion problems, and thereby improve the operational efficiency and availability of the entire network. Therefore, the significance of jitter-buffering algorithms for network environments is not limited to improving the transmission quality of WebRTC communication systems but can also contribute to the stability and reliability of the entire network.

The mean square error results of the four methods are shown in Fig. 9. The mean square error results of the four methods are shown in Figure 9. It can be seen that the algorithm used in this article has the lowest error rate in the calculation results.

Figure 8. Comparison of the results of the two algorithms in a congested network environment

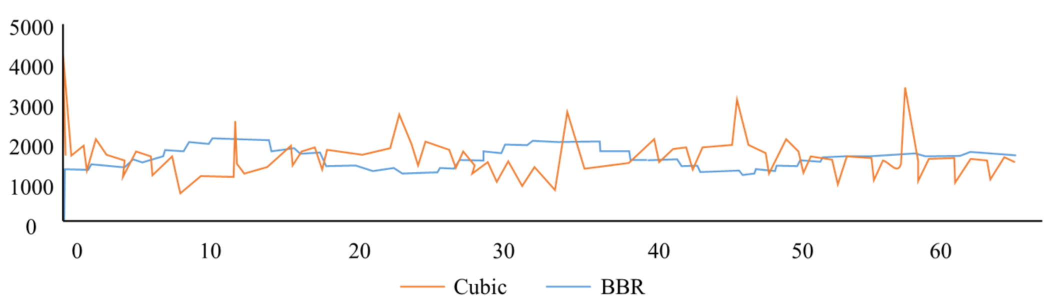
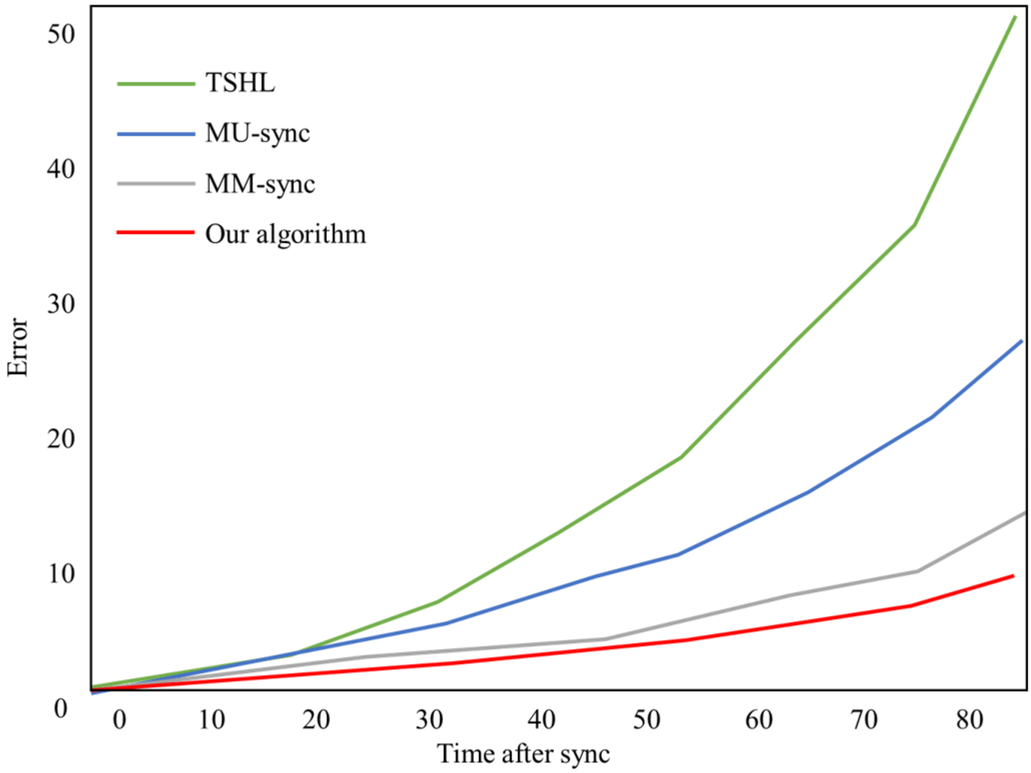


Figure 9. Comparison of the mean square error results of the four models



Analysis of Practical Applications

With the continuous development and popularization of network technology, video communication has become an indispensable part of people's daily lives and work. However, in practical applications, due to the instability and jitter of the network environment, the quality of video transmission is often difficult to ensure. Therefore, how to optimize video-transmission quality through jitter-buffering algorithms has become an important research issue. This article aims to compare the effectiveness of the jitter-buffering algorithm and the WebRTC jitter-buffering algorithm in different network environments based on a transmission-quality evaluation model, providing reference for optimizing video transmission quality.

(1) Network video transmission optimization: the transmission-quality evaluation model can help improve the quality of network video transmission, especially by comparing and selecting jitter-buffering algorithms, which can improve the stability and smoothness of video transmission.
(2) Performance optimization of communication systems: by comparing the effectiveness of different jitter-buffering algorithms in different network environments, it can provide reference for performance optimization of communication systems. Choosing a suitable jitter-buffering algorithm can improve the robustness and reliability of communication systems in various network environments.
(3) Network service quality improvement: the transmission-quality evaluation model can be used to evaluate the quality of network services and optimize them based on the evaluation results. By

applying the jitter-buffering algorithm proposed in this article, the quality of network service can be significantly improved, providing a better user experience.
(4) Network-management decision support: by using a transmission-quality evaluation model, real-time monitoring and evaluation of the network environment can be carried out, providing support for network-management decisions. Comparing the effectiveness of different jitter-buffering algorithms can help network administrators choose appropriate strategies to optimize network performance.

In summary, the research on jitter-buffering algorithms based on transmission-quality evaluation models has practical application value for optimizing network video transmission, improving network service quality, and supporting network-management decisions. The jitter-buffering algorithm proposed in this article has the following advantages compared to other methods:

(1) Based on the transmission-quality evaluation model: the algorithm in this article is based on the transmission-quality evaluation model. By evaluating and comparing the parameters of network transmission, it can more accurately reflect the quality of video transmission. Compared with traditional methods based on single indicators such as network latency or packet-loss rate, this method comprehensively considers the impact of multiple factors on video-transmission quality.
(2) Improving user experience: the algorithm proposed in this article can reduce video-transmission lag and delay by optimizing jitter-buffering strategies, thereby enhancing the user experience of watching videos. Compared to traditional fixed-buffering strategies, the algorithm proposed in this article can dynamically adjust according to the actual network conditions, making video playback smoother and more coherent.
(3) Scalability and flexibility: the algorithm proposed in this article can be combined with existing video-transmission systems and used as an independent module. It does not rely on specific network architecture or transmission protocols, has good scalability and flexibility, and is suitable for various video-transmission scenarios and applications.

In summary, the jitter-buffering algorithm proposed in this article has obvious advantages in optimizing video-transmission quality compared to other methods, based on transmission-quality evaluation models, considering different network environments, improving user experience, and having scalability and flexibility. However, this study has certain limitations.

(1) Limitations of experimental environment: this article compared the effectiveness of jitter-buffering algorithms through experiments, but only limited network environments and datasets were used. Therefore, the universality and generalization ability of the experimental results may be limited. To increase the credibility and generalization ability of the experimental results, the experimental environment can be expanded, including the use of more diverse network environments and datasets. Consider different types of networks, such as 3G, 4G, 5G, and Wi-Fi, and collect data from different regions and network conditions.
(2) Lack of comprehensive evaluation indicators: this article uses transmission quality to measure the effectiveness of algorithms, but this may not be the best indicator for comprehensively evaluating multimedia-transmission quality. Other indicators such as video quality and latency should also be considered for evaluation. In addition to the evaluation value of transmission quality, other comprehensive evaluation indicators can be considered to comprehensively evaluate the improvement effect of multimedia-transmission quality. For example, video quality evaluation metrics, latency metrics, bandwidth utilization, etc. can be considered to comprehensively measure the effectiveness of the algorithm.

(3) Limitations on the applicability of the algorithm: the jitter-buffering algorithm proposed in this article has achieved good results in experiments, but its applicability may be limited by specific network environments and transmission scenarios. The effectiveness and feasibility of this algorithm in other application scenarios still need further research and verification. In addition to the network environment and transmission scenarios used in the current experiment, the applicability of algorithms in other application scenarios can be further explored, for example, by verifying the effectiveness and feasibility in specific scenarios such as real-time video conferencing and live streaming.
(4) The trade-off between algorithm performance and efficiency: although the jitter-buffering algorithm proposed in this article has shown improvement in transmission quality, it is also necessary to consider the performance and efficiency of the algorithm. Efficient real-time transmission may be more important for certain application scenarios, while jitter-buffering algorithms may introduce additional computational overhead. In the process of algorithm design and implementation, it is necessary to balance the relationship between transmission-quality improvement and algorithm performance and efficiency. The real-time performance and efficiency of the algorithm can be improved by optimizing its complexity and reducing computational costs.

CONCLUSION

Under the influence and intervention of multimedia technology, the entire field of visual communication is undergoing tremendous changes. In order to meet the needs of users and promote social development, it is necessary to adopt more innovative ways to promote social progress. In order to improve the quality of multimedia communication, this article compares the effectiveness of the proposed jitter-buffering algorithm and the WebRTC jitter-buffering algorithm in different network environments. Compared with the original WebRTC system, the system implementing the jitter-buffering algorithm proposed in this article has improved the transmission-quality evaluation value by 19.8% in normal network environments and by 17.5% in congested network environments. The experimental results show that compared with the original WebRTC jitter-buffering algorithm, the jitter-buffering algorithm proposed in this paper has better performance in different network environments and can improve network transmission quality. This provides valuable ideas and methods for subsequent research and also helps to promote the progress and development of multimedia technology. This article compared the effectiveness of jitter-buffering algorithms through experiments, but only limited network environments and datasets were used. Therefore, the universality and generalization ability of the experimental results may be limited. In the future, to increase the credibility and generalization ability of the experimental results, it would be valuable to consider expanding the experimental environment, including using more diverse network environments and datasets—for example, considering different types of networks, such as 3G, 4G, 5G, and Wi-Fi, and collecting data from different regions and network conditions.

AUTHOR NOTE

The figures used to support the findings of this study are included in the article.

The authors declare that they have no conflicts of interest.

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REFERENCES

Ahmed, A. A., Malebary, S. J., Ali, W., & Barukab, O. M. (2023). Smart traffic shaping based on distributed reinforcement learning for multimedia streaming over 5G-VANET communication technology. *Mathematics*, *11*(3), 700. doi:10.3390/math11030700

Amiruzzaman, M., & Bhuiyan, A. (2023). A secure real-time multimedia stream data backup by network-engine for resource constrained devices. *Journal of Computing Sciences in Colleges*, *39*(3), 133–142.

Benisha, R. B. (2023). An efficient Sheep Flock Optimization-based hybrid deep RaNN for secure and enhanced video transmission quality. *Neural Computing & Applications*, *35*(11), 8065–8080. doi:10.1007/s00521-022-08083-7

Cecchinato, N., Toma, A., Drioli, C., Oliva, G., Sechi, G., & Foresti, G. L. (2023). Secure real-time multimedia data transmission from low-cost UAVs with a lightweight AES encryption. *IEEE Communications Magazine*, *61*(5), 160–165. doi:10.1109/MCOM.001.2200611

Dhar, S., Khare, A., & Singh, R. (2023). Advanced security model for multimedia data sharing in Internet of Things. *Transactions on Emerging Telecommunications Technologies*, *34*(11), e4621. doi:10.1002/ett.4621

Gaikwad, S. Y. (2024). An energy efficient weighted clustering scheme with packet service time in multimedia data transmission in WMSN. *International Journal of Intelligent Systems and Applications in Engineering*, *12*(3S), 67–79.

Janakamma, C., & Hegde, N. P. (2023). NIFR-node interference and failure recovery framework for multimedia transmission in WMSNs. *Journal of Theoretical and Applied Information Technology*, *101*(12), 5187–5193.

Jiang, L. (2023). Information visualization based on visual transmission and multimedia data fusion. *International Journal of Information Technologies and Systems Approach*, *16*(3), 1–14. doi:10.4018/IJITSA.320229

Kernen, T., & True, T. (2023). Tighter NIC/GPU integration yields next-level media processing performance. *SMPTE Motion Imaging Journal*, *132*(1), 59–68. doi:10.5594/JMI.2022.3196732

Li, S., & Li, J. (2023). Design of medical graphic aided visual elements active communication algorithm in print advertising. *Journal of Testing and Evaluation*, *51*(1), 97–106. doi:10.1520/JTE20210511

Liu, X., & Yao, R. (2023). Design of visual communication teaching system based on artificial intelligence and CAD technology. *Computer-Aided Design and Applications*, *20*(S10), 90–101. doi:10.14733/cadaps.2023.S10.90-101

Mahmoodi Khaniabadi, S., Javadpour, A., Gheisari, M., Zhang, W., Liu, Y., & Sangaiah, A. K. (2023). An intelligent sustainable efficient transmission internet protocol to switch between User Datagram Protocol and Transmission Control Protocol in IoT computing. *Expert Systems: International Journal of Knowledge Engineering and Neural Networks*, *40*(5), e13129. doi:10.1111/exsy.13129

Makeeva, E., Kochetkova, I., & Alkanhel, R. (2023). Retrial queueing system for analyzing impact of priority ultra-reliable low-latency communication transmission on enhanced mobile broadband quality of service degradation in 5G networks. *Mathematics*, *11*(18), 3925. doi:10.3390/math11183925

Mubarakali, A., Durai, A. D., Alshehri, M., AlFarraj, O., Ramakrishnan, J., & Mavaluru, D. (2023). Fog-based delay-sensitive data transmission algorithm for data forwarding and storage in cloud environment for multimedia applications. *Big Data*, *11*(2), 128–136. doi:10.1089/big.2020.0090 PMID:32673064

Okpok, M., & Kihei, B. (2023). Challenges and opportunities for multimedia transmission in vehicular ad hoc networks: A comprehensive review. *Electronics (Basel)*, *12*(20), 4310. doi:10.3390/electronics12204310

Sharma, N., Chakraborty, C., & Kumar, R. (2023). Optimized multimedia data through computationally intelligent algorithms. *Multimedia Systems*, *29*(3), 2961–2977. doi:10.1007/s00530-022-00918-6 PMID:37261261

Silpa, K., & Korra, S. (2023). Multimedia data transmission with secure routing in M-IOT-based data transmission using deep learning architecture. *Journal of Computer Allied Intelligence*, *1*(01), 1–13.

Taha, M., & Ali, A. (2023). Smart algorithm in wireless networks for video streaming based on adaptive quantization. *Concurrency and Computation*, *35*(9), e7633. doi:10.1002/cpe.7633

Viola, R., Martín, Á., Zorrilla, M., Montalbán, J., Angueira, P., & Muntean, G. M. (2023). A survey on virtual network functions for media streaming: Solutions and future challenges. *ACM Computing Surveys*, *55*(11), 1–37. doi:10.1145/3567826

Yadav, S., & Tiwari, N. (2023). Privacy preserving data sharing method for social media platforms. *PLoS One*, *18*(1), e0280182. doi:10.1371/journal.pone.0280182 PMID:36662805

Zhang, X., & Cui, Y. (2023). Analysis of application and creation skills of story-based MV micro video and big multimedia data in music communication. *International Journal of Information Technology and Web Engineering*, *18*(1), 1–17. doi:10.4018/IJITWE.325213

Zhao, S., Zeng, Z., Peng, J., & Yu, F. (2023). Achieving a secure and traceable high-definition multimedia data trading scheme based on blockchain. *Mathematics*, *11*(10), 2224. doi:10.3390/math11102224

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