Design of Adaptive Multi Stream Transmission Control Method in Social Communication Network

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Abstract. The traditional multi stream transmission control method does not allocate the transmission channel, which leads to the problem that the initial buffer loading time is too long. Therefore, this paper designs a new adaptive multi stream transmission control method for social communication networks. Build a direct communication link to obtain the received signals of relay users, build a channel bilateral matching model to allocate the transmission channels, design a multi stream distribution framework, balance the computing load of each edge server, optimize the transmission control mode, and achieve adaptive multi stream transmission control in social communication networks. Experimental results: the initial buffer loading time of the method in this paper is only 6.282 s, which proves that the application effect of the adaptive multi bit stream transmission control method in social communication networks designed this time is better.

Keywords: Social communication network · Adaptive · Multi stream transmission · Control method · Base station · Cellular user

1 Introduction

In the past decade, with the continuous development of mobile communication network and the rapid popularization of smart phones, Internet applications have penetrated into all kinds of life needs of users. Watching videos on mobile terminals through the Internet has long been an essential part of people's daily life. With the explosive growth of mobile multimedia services, text, picture, voice, video and other services have become the main body of mobile data service traffic. At the same time, with the rapid development of the Internet and the continuous popularization of high-performance mobile terminal devices, people's social scope has been expanded to the field of online social networking by using software such as making friends and instant messaging. Network and social software connect people and build a network service form with human society as the core - social network. The development of HD video technology also makes users have higher requirements for video fluency and clarity. In order to meet the increasing needs of mobile network video users, adaptive video streaming technology based on mobile terminals is also developing continuously, from the initial Microsoft smooth

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streaming technology to the current dash technology. Social networks provide a platform for mobile users to share interests, behavior dynamics, social relations and activity status. In social networks, by analyzing users' behavior, we can find some relationship between users, which can be used to describe users' choice of similar content, common interest or other similar behavior. The problem of adaptive multi code streaming transmission control in social communication networks is that HD streaming media content requires large storage capacity, large transmission bandwidth and high requirements for hardware computing power. If only upgrading the hardware equipment of the handheld terminal, it is not only difficult to support the strong business requirements, but also difficult to improve the performance of the hardware equipment to a large extent in a short time. The number of mobile network video users continues to grow rapidly, and the challenges related to device heterogeneity and network heterogeneity that adaptive video streaming technology needs to deal with are also more severe. In some hotspot areas, users are densely distributed, and multiple users may request the base station to download the same content. In the traditional communication mode, the base station needs to repeatedly send this content to these cellular users, but this will greatly increase the burden of the base station. In this case, using social traffic information network to push or share content among users with close social relationship has become a feasible method to reduce the burden of base stations. On the other hand, for video service providers, it is necessary to build a large-scale online video service platform to meet the demand of massive high-definition video. It is necessary to ensure the high reliability of the service and the scalability of the scale, which undoubtedly brings huge capital pressure to video service providers.

Relevant scholars have made some progress in the research of multi stream transmission control. For example, Luo Zhiwei et al. Proposed an embedded multi-channel wireless video transmission rate adaptive algorithm [1]. The dm368 chip acquires multi-channel wireless video data, calculates the video data transmission rate according to the Gaussian function, and realizes the bit rate equalization control of multi-channel wireless video transmission rate through the extremum suppression method. This method can improve the video transmission rate, and this method can improve the buffer capacity of wireless video transmission, But the delay jitter is high. Shao Ruirui et al. Proposed a measurement method of multi stream transmission rate of communication network based on toughness [2], constructed a toughness function to obtain multi stream transmission rate data, and realized multi stream transmission channel selection through transmission rate matching method, which effectively improved the accuracy of multi stream transmission rate measurement of communication network, but there was a problem of too long buffer loading time.

In view of the above problems, this paper designs an adaptive multi stream transmission control method for social communication networks.

2 Design of Adaptive Multi Stream Transmission Control Method in Social Communication Network

2.1 Prediction of Social Communication Network Bandwidth

The cellular communication and communication hybrid network architecture of social communication network is divided into two layers: social layer and physical layer [3]. In the social layer, the dynamic behavior of users on the social platform can be used to reflect the social connections between users. Therefore, users' behavior on social platforms such as microblog and twitter can be used to find the social relationship between users. Communication technology is a new communication technology that allows terminal equipment to directly communicate by reusing the spectrum resources of traditional cellular users in the cell under the control of the base station in the cellular system. It can not only improve the spectrum efficiency of cellular communication system and reduce the transmission power of terminal, but also solve the problem of spectrum resource shortage in wireless communication system. The direct communication link is built in the physical layer, and whether it can be built successfully mainly depends on the transmission distance between two mobile user devices. Each user in the social layer corresponds to a terminal mobile user in the physical layer. In order to realize information push or content sharing through communication links, it is necessary to comprehensively consider the social layer and physical layer information of the system [4, 5]. As we all know, WiFi technology or Bluetooth technology has strong interference with each other when users use it due to the use of unauthorized frequency bands, which will reduce the user experience. Different from the unauthorized frequency band used by WiFi technology or Bluetooth technology, the communication technology can use the authorized frequency band. When the communication technology uses the authorized frequency band to enable the two devices to communicate directly, the base station can assist in the coordination of interference. In channel modeling, Rayleigh fading is used to model small-scale fading, and free space propagation path loss is used to model large-scale fading. The received signals received by the receiving user in the transmission link and the base station in the cellular transmission link are:

$$E_0 = \frac{w \times \alpha^{-1}}{\beta} \times Z^2 \tag{1}$$

$$E_1 = \frac{w \times \frac{1}{\alpha}}{\beta} \times Z \tag{2}$$

In formulas (1) and (2), w represents the transmission power of the transmitting user, α represents the transmission power of the cellular user, β represents the channel response of the communication link, and Z represents the channel response of the cellular communication link. In general, if there is a strong social relationship between two users, the probability of establishing a direct communication link between them for content sharing will be higher, because their content preferences are more similar than two users with weak social relationships. In addition, having better channel conditions between two mobile user devices will promote the construction of direct communication links. In the physical layer, if the distance between two mobile users is within the transmission range required by the communication link construction, and the service quality of cellular users and communication users can be guaranteed, the communication link can be built smoothly. In this system, communication users can share the uplink spectrum resource blocks occupied by cellular users, and each cellular user can only occupy one spectrum resource block, and each resource block can only be multiplexed by one communication user at most, and vice versa. How to establish, continue and end the communication should be controlled by the base station. Communication from establishment to completion can be understood as a process in which users request resources from the cellular system, the base station allocates resources to communication users, the base station maintains the transmission service of communication, and the cellular system recovers spectrum resources. When the bandwidth formed by the transmitting user and the receiving user affects the multiplexing resource block of the communication link, the signal to interference plus noise ratio received by the receiving user and the received by the base station are respectively expressed as:

$$G = \frac{1}{l} \sum H_{\delta}^{2} - H^{\phi} \tag{3}$$

$$G' = \sum \left| \frac{l - H^{\phi}}{H_{\delta}^{2}} \right| \tag{4}$$

In formulas (3) and (4), l represents the channel response of the interference link between the cellular user and the receiving user, H represents the channel response of the interference link between the transmitting user and the base station, δ represents the free space path loss factor, and ϕ represents the Rayleigh channel factor subject to complex Gaussian distribution. The multiplexing of uplink spectrum resources leads to the common channel interference between the base station and the receiving user. Therefore, the communication quality of cellular communication link and communication link is affected. In the social layer, the higher the similarity of two users' preferences for content, the closer the social relationship between users. However, due to the uncontrollability and uncertainty of users' social behavior, it is difficult to find an appropriate model to describe the characteristics of users' behavior. The user can also obtain the media content from the adjacent user terminal that has obtained the media service with the help of the communication network, so as to alleviate the downlink transmission pressure of the operator's cellular network. In addition, the traditional cellular communication between short-range users can also be switched to the communication mode to realize the unloading of cellular network traffic. Therefore, this section uses the probability of users selecting similar content to represent the similarity of users' social behavior. The higher the normalized correlation of the probability of users selecting similar content, the closer the social relationship between users. Communication technology can not only provide adaptive data service services, especially for data sharing and transmission services in local user communication services, but also flexibly meet business needs in terms of file sharing, information sharing and other services. In order to obtain the probability distribution of users choosing similar content, the system integrates the historical records of users' behavior on different social platforms, and obtains the probability density function of users choosing similar content by using Bayesian nonparametric model, so as to obtain the social relationship strength between users.

2.2 Build Channel Bilateral Matching Model

2.2.1 Meaning of Bilateral Matching

Bilateral matching means that the matching parties are two sets without intersection, and the elements of both parties are matched according to the preferences of each element, so that the elements in the two sets are related to each other on the basis of meeting a certain stability principle. The essence of matching is actually bilateral exchange. Therefore, both sides of matching will have their own sensitive preference list. Relay communication technology can be divided into single hop communication and multi hop communication. Single hop relay cooperation means that the destination can be reached through one relay cooperative communication from the source, while multi hop cooperative communication is the communication of nodes that can reach the destination through multiple relay cooperation. It mainly includes the following four models: single relay double hop model, single hop parallel model, multi hop serial model and multi hop cooperative model. According to the stable existence of bilateral matching, the matching problems can be divided into three categories: one-to-one matching, many to one matching and many to many matching. One to one matching means that each element in the two matching sets can match at most one element in the other set. From the perspective of signaling exchange and data exchange, the user still maintains the signaling link with the base station, which is no different from the traditional cellular user. The base station still allocates the wireless spectrum resources and manages the traffic of communication users, that is, traffic billing, mobility and security management. Many to one matching means that in one of the matching sets, at least one element can match multiple elements in the other set, while in another set, each element can match at most one element in the other set. For example, for school enrollment, a school can recruit multiple students, while a student can only select one school. Many to many matching means that at least one element in the two matching sets can match multiple elements in the other set.

2.2.2 Construction of Channel Bilateral Matching Model

In the first hop communication of cooperative communication, the signal size received by the relay user is:

$$R = \varepsilon \frac{1}{\eta} + \sqrt{F_2^{\eta-1} - r} \tag{5}$$

In formula (5), ε represents the fading coefficient, η represents additive Gaussian white noise, F represents the user distance, and r represents the distance between the user and the base station. However, different from traditional cellular users, the data transmission link between communication users does not need to be forwarded by the base station, but directly establishes a data transmission channel between two users.

This way of communication can not only get faster transmission rate and better user experience, but also reduce the burden of the base station to a certain extent. For example, some popular contents do not have to be downloaded from the base station repeatedly. Some previous work combining social network and communication technology focused on quickly sharing the same content to multiple users in the region, such as shortening the transmission time of the content in the whole region as much as possible. However, how to share and push the content when users have different preferences for the content has not been studied in detail. Social network is a new communication medium closely related to distance. It is a new communication service based on direct data transmission of short-range communication [6, 7]. Communication should meet a variety of different forms of service requirements. For example, communication technology can often be applied to a variety of local communication services, including communication in a small outdoor communication environment or indoor communication. Assuming the application scenario of a concert, in this case, the video service, especially the welcome of the audience, the concert organizer can apply for the communication spectrum resources from the cellular system, and then the audience participating in the concert can download the video service provided by the concert to each other by using the communication technology. In this way, it can not only meet the requirements of listeners, but also effectively reduce the load of base stations in cellular cells. For transmitting users and receiving users, the closeness of their social relationship is determined by the normalized correlation of their probability of selecting similar content, which is expressed as:

$$W_{pq} = \frac{(corr|p, q| - \eta)}{2} \tag{6}$$

In formula (6), p represents the transmitting user and q represents the receiving user. In order to use communication technology to push or share content among users with close social relationships. Firstly, social relations reflect the consistency of users' love for similar content. Using social relations can determine the transmitting users and receiving users of communication. This process can be regarded as a user discovery process. While communicating, the system can provide traffic and data sharing services of the Internet. With the increasing popularity of high-definition video and other media services, its large traffic characteristics also bring great challenges to operators' core network and spectrum resources. Secondly, communication users need to reuse the spectrum resources occupied by cellular users for communication, and the resulting co channel interference can not be ignored. Therefore, it is necessary to design an effective resource management scheme, so as to improve the system performance while ensuring the user communication quality. Using the local characteristics of communication, local media services can save the spectrum resources of the operator's core network. In the hot area, operators or content providers can deploy media servers to store the current popular media services in the media server, while the media server provides corresponding media services to users with business needs in an integrated mode. Based on the above two aspects, an effective content distribution scheme is proposed for cellular communication and communication hybrid networks, comprehensively considering the information of social layer and physical layer of the system, and optimizing user discovery and resource allocation, that is, the matching among users, shared content and spectrum resources, While improving users' satisfaction with the received distribution content, it can improve the performance of the system through an appropriate spectrum resource allocation scheme.

2.3 Design of Multi Stream Distribution Framework

Dynamic rate adaptive transmission control requires the base station or user terminal equipment to dynamically decide the video rate version sent to the user according to the dynamic network channel conditions and user buffer status, so as to maximize the user's viewing experience. In the wireless operator network, there is a scenario where multiple users request the same video URL at the same time. In this scenario, the dash transmission scheme considers that the "multicast" transmission mode of MBMS can be used for service transmission. We model the joint edge caching, transcoding and distribution decision-making problem and form it into a multivariable nonlinear integer programming problem, which aims to minimize the operating cost of the edge network and balance the computing load of each edge server under the constraints of the storage and computing resource capacity of the edge server. MBMS carries out the "multicast" transmission of dash content through MBSFN transmission mode. MBSFN mode has the following advantages: because the mobile terminal can effectively use the signal energy from multiple cells, the received signal strength of users can be effectively improved under MBSFN mode, especially the signal strength of users in edge cells in MBSFN area. Transmitting video with high-precision and panoramic characteristics brings a great burden to the existing network, including bandwidth resource consumption, storage resource consumption of relay cache, transmission delay, etc. [8–10]. Reduce signal interference, especially for users at the boundary of different cells. Because in the MBSFN transmission mode, the signal received by the user from the adjacent cell will be useful and will no longer be regarded as the interference signal of wireless transmission as in unicast. However, on the one hand, when users watch video, they will only watch the FOV area, so they can only transmit the video content in the FOV area or only keep the video content in the FOV area in a high bit rate version. On the other hand, edge caching and edge computing technology can effectively alleviate the transmission pressure of the return link, and the two resources can cooperate to share the video relay pressure. The diversity of wireless channel fading, because information can be received from several geographically separated places, usually the channel looks like time division multiplexing or frequency division multiplexing as a whole. In order to realize the above transmission framework, this chapter further optimizes the functional modules of the edge server. The execution edge server consists of four modules, as shown in Fig. 1:

According to Fig. 1, the functional modules of the edge server include: request collection module, request prediction module, cache decision module and distribution decision module. The request collection module is responsible for collecting the video slice requests of each user at the beginning of each transcoding and distribution stage. At the beginning of each cache stage, the request prediction module retrieves the user's historical request information from the request collection module and analyzes it to predict the video slice request of each user in this cache cycle time. The transcoding server is set up at the edge of the network (base station). For each video stream, it only needs to transmit a code rate version of the video stream from the core network to the base station once. After that, the transcoding server at the base station transcodes the

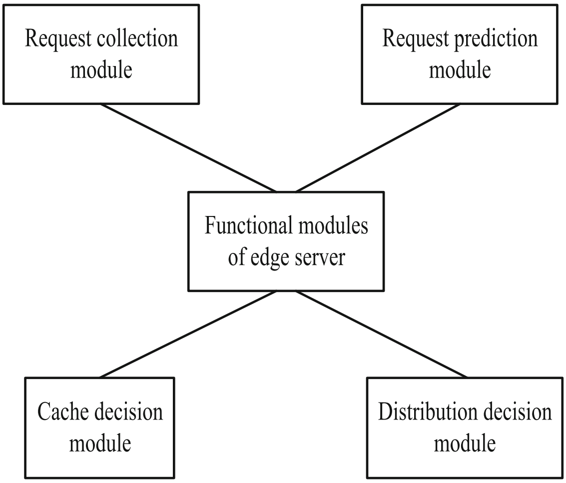

Fig. 1. Implementation of edge server function module

best code rate set by obtaining the user's demand and channel status in real time, which not only reduces the bandwidth consumption of the core network, but also improves the adaptability, system performance and user QoE of the system. At the beginning of each caching stage, the caching and distribution decision module makes a joint caching decision according to the predicted user request information, network topology information and the storage and computing resource capacity information of each execution edge server. At the same time, the edge server also has the function of wireless spectrum (bandwidth) adjustment, which can improve the bandwidth utilization by adjusting the user's bandwidth. The bandwidth adjustment module can also transcode a new code rate set again in combination with the transcoding server to further improve the system performance. At the same time, in each transcoding and distribution stage, the module makes transcoding and distribution decisions according to the real-time collected user request information and the current cache state of each execution edge server. Each execution edge server includes a cache module, a transcoding module and a distribution module. It is responsible for the caching, transcoding and distribution of video slices according to the decision command of the decision edge server. The bandwidth adjustment module is responsible for readjusting the bandwidth resources of users. While improving the bandwidth utilization, it can transcode a new code rate set in combination with the transcoding server to further improve the performance of the control method.

2.4 Optimize Transmission Control Mode

Combining adaptive streaming, edge caching and edge computing technology based on tile granularity, we can make full use of the local viewing characteristics of video and comprehensively utilize the multiple resources in the edge network to support the adaptive streaming of video with the minimum operation cost of the edge network. Adaptive transmission control technology includes two different driving modes, client-side driving and server-side driving. Client driven adaptive transmission control is also known as pull streaming media technology, and server driven adaptive transmission control is also known as push streaming media technology. An edge caching, transcoding and distribution framework for video adaptive streaming is designed, and the optimization method is used to model and solve the joint decision-making problem of caching, transcoding and distribution by using the storage and computing resources of the edge network in tile granularity coordination, so as to minimize the comprehensive operation cost of the edge network under the constraints of network resource capacity and ensuring user request response. In the adaptive transmission control driven by server, the most widely used is the real-time transmission protocol. RTP runs on the user datagram protocol, and UDP does not contain any rate control mechanism. This makes RTP more suitable for low latency and best effort streaming media transmission. In the usual server-side driven adaptive transmission control, the server uses the encoding rate as the transmission rate to match the consumption rate of the client. The expression formula of the effect function of maximizing users is obtained as follows:

$$V_g = \frac{|u - s|}{g} \times u^{-1} \tag{7}$$

In formula (7), u represents the number of code rate jitters, s represents the stability parameter, and g represents the penalty factor. In order to make full use of the storage and computing resources in the edge network to assist the video transmission, we need to consider not only the cooperation between multiple edge servers, but also the coupling and cooperation between multiple tasks (caching, transcoding and distribution). Considering that users directly obtain video slices from the remote server through the return link will cause large transmission delay, our proposed transmission framework follows the principle of allowing all users to obtain the requested video slices only from the edge server. This can ensure that the size of the client cache remains constant for a certain period of time, and make the best use of network resources. However, if the packet is lost or the network transmission is delayed, the recovery rate of the client packet will be lower than the consumption rate, resulting in buffer overflow and playback interruption. Here, adaptive transmission control technology can solve this problem well. In order to prevent buffer overflow, the server will automatically choose to send media streams with low bit rate. Moreover, in order to meet the feasibility of transcoding, we assume that the edge server already holds the highest bit rate version of each video slice. In the caching stage, the decision edge server makes the optimal caching decision by analyzing the storage and computing resource capacity and user request characteristics of each execution edge server, and sends the decision control information to each execution edge server. In this way, the media consumption rate of the client can be reduced, so as to offset the impact of the reduction of network bandwidth. When the network environment improves, the server will automatically choose to send media streams with high bit rate instead of before. By automatically monitoring the available bandwidth and buffer area and adjusting the transmission rate by selecting media streams with different bit rates, push streaming media transmission realizes the smooth playback of video with the highest quality level as possible. When the client plays video clips one by one, it can seamlessly reconstruct the original media stream. Each execution edge server downloads the video slices to be cached from the remote server according to the received cache decision control information. In the transcoding and distribution stage, the decision edge server decides the optimal edge server user connection pair (distribution decision) according to the cache content, calculation resource capacity and user request information of each current execution edge server. In the download process, the client automatically selects the appropriate bit rate video fragment according to the current available bandwidth. In this way, the client realizes adaptive transmission based on available bandwidth. Assuming that for any user, any dash video program can only be transmitted with one code rate value in one transmission mode at the same time, the expression formula of code rate transmission is:

$$T = \sum_{x=1} \frac{|x-y|^2}{M_{xy}} \times \sigma_y \tag{8}$$

In formula (8), x represents the dash video set transmitted in PTP mode, y represents the dash video set transmitted in SFN mode, M represents the total resource consumption, and σ represents the number of resource blocks. Although the partition structure of video files in different formats is different, the basic principle of partition is the same. When the audio data and video data are not interleaved, usually the audio frame is composed of some audio samples with the same time length, and each audio frame can be decoded separately by the audio codec. The execution edge server establishes a transmission connection with the user according to the distribution decision information, and distributes the requested video slices to the user. Specifically, for a specific user, if the execution edge server connected to it has cached the requested video slice, it will directly obtain the cached video slice; otherwise, the execution edge server connected to it will transcode the requested video slice using computing resources and transmit it to the user. Therefore, only a certain number of audio frames need to be combined into audio data with the same length of segmentation and filled in the segmentation. The processing of video data is completely different, because video frames cannot be decoded separately. Therefore, video partition exists in the form of picture group in segmentation.

3 Experimental Test

3.1 Experimental Preparation

This paper constructs the system in a LAN. The system consists of source, server and client. The client controls the network bandwidth between the server and the client through NetLimiter software. The slices in this paper adopt three quality levels: 350 kps, 750 kps and 1500 kps. Each slice is about 3–7 s long. Based on evalvid's open source video streaming architecture, rewrite the terminal and server-side programs. The traditional dash rate control algorithm and eams cloud algorithm are added to the terminal and server respectively. Add the energy model of NS3 to the terminal node to simulate the real-time energy consumption of the terminal equipment. The minimum playback time allowed for the media stream in the client cache is 3800 ms. After the system is started, it runs stably. We analyze the 1200 pieces sent.

4 Experimental Result

At the same time, the radius of the cell is set to 1200 m, and the total bandwidth of the system is 60 MHz. The adaptive multi stream transmission control method of social communication network based on cloud computing and the adaptive multi stream transmission control method of social communication network based on spatial Poisson point are selected for experimental comparison with the adaptive multi stream transmission control method of social communication network in this paper. Under different bandwidth conditions, the initial buffer loading time of the three methods is tested. The less the time, the better the performance is proved, The experimental results are shown in Table 1, 2, 3, 4 and 5:

Table 1. Initial buffer loading time with bandwidth of 500 kbps (s)

Number of experiments	Adaptive multi code stream transmission control method of social communication network based on cloud computing	Adaptive multi code stream transmission control method of social communication network based on spatial Poisson point	Adaptive multi code stream transmission control method for social communication network in this paper
1	1.998	2.765	1.233
2	2.342	1.276	1.098
3	1.672	2.868	0.868
4	1.801	1.766	0.965
5	2.553	2.671	0.575
6	1.166	2.337	1.232
7	1.869	1.976	1.004
8	2.224	2.117	1.673
9	1.867	1.988	0.984
10	2.653	2.202	1.223

It can be seen from Table 1 that the average initial buffer loading time of the social communication network adaptive multi stream transmission control method in this paper and the other two social communication network adaptive multi stream transmission control methods are 1.086 s, 2.015 s and 2.197 s respectively.

It can be seen from Table 2 that the average initial buffer loading time of the social communication network adaptive multi stream transmission control method in this paper and the other two social communication network adaptive multi stream transmission control methods are 2.141 s, 4.089 s and 3.809 s respectively.

It can be seen from Table 3 that the average initial buffer loading time of the social communication network adaptive multi stream transmission control method in this paper and the other two social communication network adaptive multi stream transmission control methods are 4.237 s, 6.868 s and 7.053 s respectively.

Table 2. Initial buffer loading time with bandwidth of 1000 kbps (s)

Number of experiments	Adaptive multi code stream transmission control method of social communication network based on cloud computing	Adaptive multi code stream transmission control method of social communication network based on spatial Poisson point	Adaptive multi code stream transmission control method for social communication network in this paper
1	4.979	3.099	2.099
2	4.883	3.234	2.356
3	4.652	4.543	2.673
4	3.562	3.742	1.977
5	3.867	4.987	2.390
6	4.456	3.234	1.793
7	3.982	4.549	2.212
8	4.459	3.339	1.994
9	3.030	3.320	1.868
10	3.018	4.038	2.052

Table 3. Initial buffer loading time with bandwidth of 1500 kbps (s)

Number of experiments	Adaptive multi code stream transmission control method of social communication network based on cloud computing	Adaptive multi code stream transmission control method of social communication network based on spatial Poisson point	Adaptive multi code stream transmission control method for social communication network in this paper
1	6.877	6.424	4.383
2	6.678	7.362	3.666
3	7.334	6.988	3.453
4	6.567	7.211	5.122
5	7.122	6.874	3.868
6	6.776	7.477	4.019
7	7.289	6.738	3.674
8	6.577	7.223	4.433
9	7.334	7.236	5.020
10	6.123	6.994	4.728

It can be seen from Table 4 that the average initial buffer loading time of the social communication network adaptive multi stream transmission control method in this paper

Table 4. Initial buffer loading time with bandwidth of 2000 kbps (s)

Number of experiments	Adaptive multi code stream transmission control method of social communication network based on cloud computing	Adaptive multi code stream transmission control method of social communication network based on spatial Poisson point	Adaptive multi code stream transmission control method for social communication network in this paper
1	15.567	14.277	9.004
2	14.673	15.938	9.863
3	13.535	14.261	10.028
4	14.679	13.304	9.378
5	13.464	13.346	8.433
6	14.266	14.099	8.646
7	14.309	13.017	9.893
8	15.206	13.048	10.017
9	14.217	14.183	9.363
10	15.236	14.125	10.738

and the other two social communication network adaptive multi stream transmission control methods are 9.536 s, 14.515 s and 13.960 s respectively.

It can be seen from Table 5 that the average initial buffer loading time of the social communication network adaptive multi stream transmission control method and the other two social communication network adaptive multi stream transmission control methods are 14.409 s, 19.774 s and 19.931 s respectively.

In order to further verify the effect of adaptive multi code stream transmission control in social communication networks of this method, cloud computing method, spatial Poisson point method and this method are used to verify the accuracy of multi code stream transmission control, and the results are shown in Fig. 2.

According to the analysis of Fig. 2, when the data volume is 100 GB, the multi stream transmission control accuracy of the cloud computing method is 70%, the multi stream transmission control accuracy of the spatial Poisson point method is 56%, and the multi stream transmission control accuracy of the method in this paper is 96%; When the data volume is 500 GB, the multi stream transmission control accuracy of cloud computing method is 74%, the multi stream transmission control accuracy of spatial Poisson point method is 63%, and the multi stream transmission control accuracy of this method is 98%; The multi stream transmission control accuracy of this method is always high, which shows that this method can improve the multi stream transmission control effect.

Table 5. Initial buffer loading time of 2500 kbps bandwidth (s)

Number of experiments	Adaptive multi code stream transmission control method of social communication network based on cloud computing	Adaptive multi code stream transmission control method of social communication network based on spatial Poisson point	Adaptive multi code stream transmission control method for social communication network in this paper
1	21.099	19.376	14.231
2	19.271	21.297	13.565
3	18.468	18.013	15.786
4	21.091	20.248	13.908
5	18.286	21.317	14.122
6	18.370	18.721	12.453
7	21.265	20.404	15.904
8	20.313	21.892	14.436
9	18.208	18.274	14.560
10	21.371	19.763	15.121

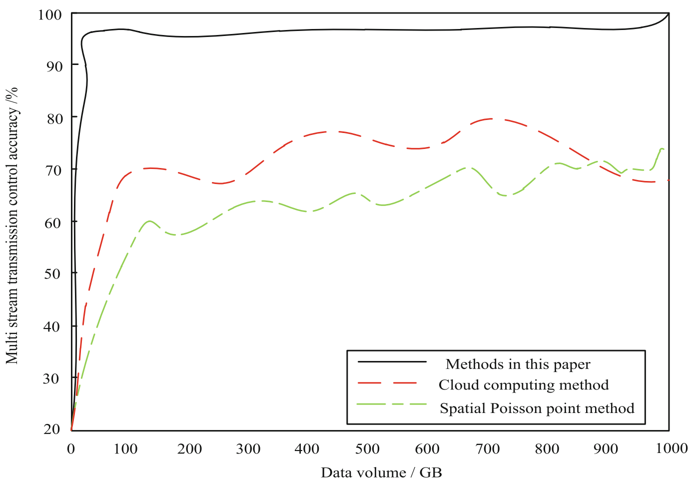

Fig. 2. Control accuracy of multi code stream transmission

5 Concluding Remarks

The adaptive multi stream transmission control method of social communication network in this paper can better adapt to channel changes and improve bandwidth utilization, and further improve system performance by adjusting user bandwidth. In this paper, the problem of adaptive video streaming is further divided into a problem of bandwidth and code rate adjustment. Through analysis, the rate allocation sub problem is transformed into user grouping problem, and it is proved that the user grouping problem is NP hard. According to the analysis of the energy consumption of smart phones in each link of video streaming processing, the mathematical model between the power consumption factors of smart terminal equipment and video codec parameters is established, and the improved adaptive algorithm of code rate and resolution based on energy consumption perception is used to solve it, which improves the performance of adaptive multi code streaming transmission control method in social communication network.

Although the method in this paper achieves good initial buffer loading efficiency, it still has high algorithm complexity. Therefore, the next research direction is how to reduce the complexity of the algorithm and improve the loading efficiency.

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