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Research on an End-to-end Service Quality Assurance Algorithm of Application Layer Applied in Mobile Client Multimedia Conference

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Abstract: Since the long delay of one-way transmission of wireless network leads the multimedia video to frame loss, frame skip, interruption and other issues, this paper presents an end-to-end service quality assurance algorithm of application layer. For the characteristics of bandwidth limitations, sensitive delay and channel status about multimedia video, this algorithm combines the subsection and restructuring of data packet with heterogeneous transmission of path diversity about wireless network, segment and transmit the data packet in considering the bandwidth, delay, length and weighting and other parameters. Each subsection reaches the client after the transmission of path diversity, and then does validity checking and bandwidth limitation restructuring. Simulation experiment shows that: this algorithm can effectively integrate heterogeneous bandwidth of wireless network, and reduce the one-way delay of video streaming transmission.

Keywords: Wireless Network; Multimedia; Quality Assurance; Path Diversity Transmission

1. Introduction

Mobile multimedia conference means in a heterogeneous wireless network environment, users can access conference system through a variety of smart mobile terminals (such as laptops, smart phones, tablet PCs, etc.), and do comprehensive information exchange [1-3]. This information presents synchronously or asynchronously multimedia data such as real-time audio, video, text, images and so on. Typical applications are video conference, distance learning, online multiplayer games. Compared with the traditional multimedia conference system, its advantages are: (a) Mobility. Solved the problems that traditional multimedia conference terminal lacks mobility support, and limiting users can only have conferences in wired and fixed network environment; (b) Economy. Mobile multimedia conference does not need to use dedicated hardware devices and communication networks, which largely reduces the user's cost, and make the popularity of mobile multimedia conference becomes possible; (c) Compatibility. Be able to support multiple types of wireless network, mobile terminal, operating system, network protocols and other related hardware and software. With the rapid development of heterogeneous wireless network and smart mobile terminal, mobile multimedia conference gradually become a new trend and research focus [4-6]. Heterogeneous wireless network is the collection of bandwidth, signal range, network protocols, application scenarios and other wireless networks with various features, such as cellular networks and various kinds of digital video broadcast network. In the future of fourth-generation communication systems, there will be more wireless networks which play an important role in different applications. The current research focus of heterogeneous wireless network is using characteristics between varieties of wireless networks for complement, and to provide users with better quality of service [7]. For example, combine the characteristic of mobility support of cellular networks with the low-cost and high-bandwidth of wireless local area network, is used to support the uplink and downlink transmission of mobile multimedia conference video, audio, images, text and other data streams, not only meet the user's needs to exchange comprehensive information, but also save costs. The application scene of mobile multimedia conference in the environment of heterogeneous wireless network is shown in Figure 1.

However, guarantee the quality of service for multimedia conference in unreliable, changing environment of wireless network, is facing serious challenges, including: (a) Bandwidth limitations. Although the problem of bandwidth limitation has been eased up in recent years, but in practical applications, the available bandwidth for users is still very limited. (b) Sensitivity delay. End to end one-way delay is an important indicator to measure the quality of service of real-time interactive multimedia application. (c) Channel state. The channel state of wireless network is complex and changeable, which was demonstrat-

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ed by a high error rate, packet loss rate, bandwidth fluctuations, delay jitter, etc., to solve such problems, it requires the use of feedback retransmission, forward error correction algorithm, etc., which will further increase the delay in a certain extent [7-10].

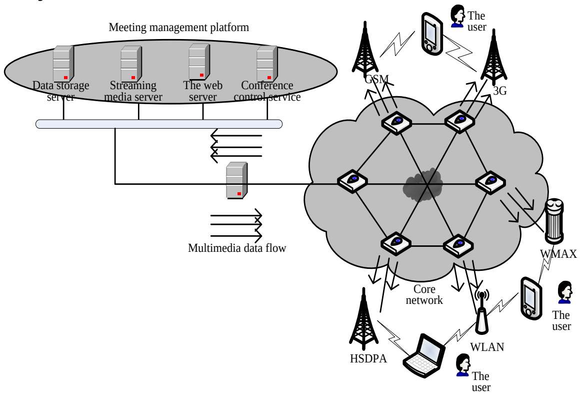


Figure 1. The application scene of mobile multimedia conference in the environment of heterogeneous wireless network

Therefore, in most cases, relying on a single wireless network can not support mobile multimedia video transmission which needs low latency, high bandwidth. Solving this problem requires the synergic transmission between heterogeneous wireless networks, and the research work has been divided into two directions: Vertical switching and path diversity.

Path diversity technique is an efficient algorithm to solve the guarantee problem of service quality of multimedia conference in previous text. In this research field, there has been some related work. Chebrolu and Rao proposed EDPF (Earliest Delivery Path First) algorithm, the idea is when sent each packet, select the shortest path of end-to-end delay from a variety of available wireless networks [11-12]. This algorithm reduces the total delay of video streaming to some extent, but because there is a single path bandwidth limitations and did not effectively deal with the data packet whose length is too long, one-way delay problem has not been effectively resolved. Jurca and some other people design LBA (Load Balancing Algorithm) algorithm based on principle of load balancing, which fully consider the network bandwidth, play delay, data packet weight and other factors. Nightingale and some other people present OPSSA algorithm regarding selecting and switching problem between multi-paths in mobile network [13]. Han and some other people design and implement a virtual path build system from end to end based on fountain code, and the main purpose of this algorithm is to solve the problem of packet loss and bandwidth fluctuations. The above research rationally and effectively uses the path diversity transmission to solve the problems faced, but there is no effective algorithm for one-way delay. This is because the main function of path diversity is to solve the problem of limited bandwidth of a single wireless network, but the bandwidth is one of the important factors to affect one-way delay. According to ITU TG [14]. 114 standard, the length of data packet is an important factor as well, the data packet with long length will cause frame dropping, skipping, intermittent problems due to the long delay, and seriously affect the quality of service and the user's experience. Therefore, to solve this problem, it needs effective adjust to the length of packet. Currently relative research work has been done regarding the length of data packet, and mainly concentrated in the MAC layer. After study, Lin and some other people found the relationship between frame size and the throughput of WLAN, they use neural network algorithm and consider a variety of factors, calculate the frame size and adjust self-adaption, then improve network throughput. Modiano found that the frame size which is too large is not suitable for transmission in error-prone channel, and elicited the calculation formula between optimal size of the frame and the channel error rate. Therefore, this article uses path diversity transmission to solve bandwidth limitation of a single wireless network, and use segmentation reorganization algorithm to adjust the size of data packet effectively, and through server client forms on application layer with multi-path data packet scheduling algorithm to combine these two algorithms and fulfill them. The reason for selecting the application layer is the underlying parameters and protocol are controlled by integration service providers and hardware manufacturers, it is difficult to coordinate and unify, and the application layer parameters are within the controlled range of the application service provider, optimize and adjust it according to specific needs.

2. Service Quality Guarantee Algorithm

Overall structure of fast transmission about End-to-end virtual access guarantees multimedia video stream is shown in Figure 2, and this algorithm comes true by the packet scheduler of the server, and the packet receiver of the client. The scheduler of server uses the Packet Fragmentation and Delivery Algorithm (PFDA), and Packet Receive and Reassembly Algorithm (PRRA). The above process can be divided into the following specific steps: (1) fragment the packet and transmit; (2) build end to end virtual path for path diversity transmission; (3) packet reception and restructuring.

2.1. Packet Fragmentation and Reassembly

The video packet described in this paper is the network abstraction layer unit (NALU) in H. 264 standard. In H. 264 standard, treat NALU as a unit to support the transmission of code data in network.

The length of NAL changes according to the difference of coding parameter and video test sequence, and the main parameter is bit rate. In this experiment, the standard YUV video sequences Foreman-cif respectively encode using 1Mbps and 3Mbps bit rate, and the other parameters are the same, and then calculate the generated NAL unit. The percentage (probability density) about the total length of length distribution and the length of video stream data is shown in Figure 3.

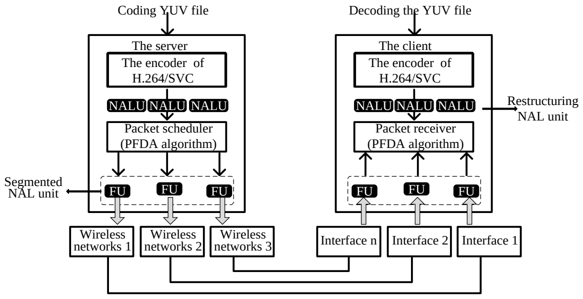


Figure 2. Overall architecture

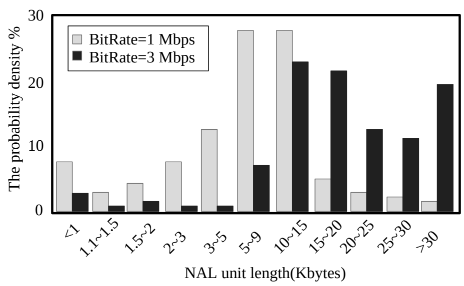


Figure 3. NAL cell length distribution diagram

As we can see from the figure, when the bit rate is larger, the quantity of NAL unit with greater length generated more. According to information theory, bit rate is the necessary condition to guarantee the HD quality and minimal distortion for video. So if you want the video distortion is minimized, you need to consider the delay of the packet whose length is too large, otherwise the video intermittent, dropped frames, skipping phenomenon is very obvious. In the actual available bandwidth limited condition, relying on a single wireless network, in most cases, can not be control one-way delay within the tolerance range, so this paper uses segmentation recombinant algorithm of the NAL unit to make effective adjust on length.

The segmentation and reassembly of NAL unit has been described in the RTP specification of H. 264/AVC and H. 264/SVC, but its main purpose is to consider the maximum transmission unit (MTU) length, unequal error protection, and the receiving end cache capacity and other issues, and has not been used in combining with transmission of path diversity. This paper follows the above two specifications, according to the specific application needs, use custom headers to segment reorganization, and the specific process is shown in Figure 4.

Set length of NAL unit which wait for fragmentation as Luo bytes, and divide it into two lengths, one is Luo byte and the other is byte, and adds a 16-byte header in these two parts. The information contained in header respectively are code number belongs to NAL unit, the total length belongs to the NAL unit, the starting position of segment and length of the segment, each occupy 4 bytes. The header information of those two sections is shown in Figure 4, so we can analogy the process that single NAL unit can be divided into multiple sections.

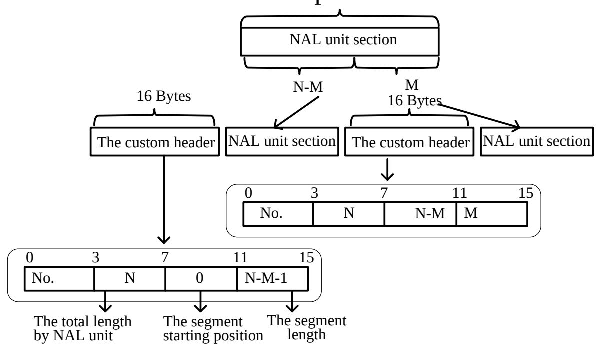


Figure 4. Diagram for NAL unit segmentation process

While restructuring the client, when receiving the first segmentation, apply for a memory space of length N, and copy the segment according to starting position and length, continue to wait to receive packets, when receiving the second segment and copied to the applied memory space, the restructuring of this NAL unit is finished. Restructuring process is shown in Figure 5: Refer to the factors which need to be considered for NAL unit segmentation restructuring and implementation process will be described in details in Section 3 PFDA-PRRA algorithm. Because the work in this paper is in the application layer, below this layer, the further packing, splitting, etc for NAL unit is performed by the underlying system and related agreements, in order to unify the name, NAL unit is packet which described in the previous and late text.

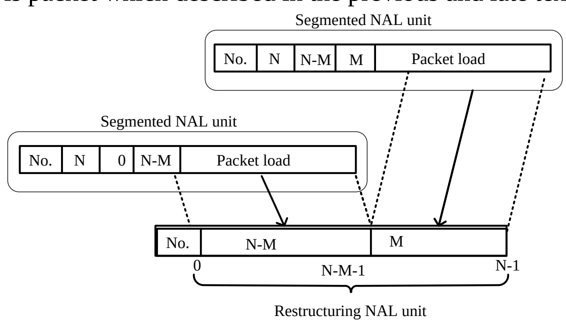


Figure 5. Diagram about NAL unit restructuring process

2.2. Path Diversity Transmission

Path diversity transmission means data transmission in the switching network that from the source node to the destination node via two or more paths to transmit at the same time. The path can be expressed as a pair of source destination addresses to reach a single destination node. When there are multiple parallel transmission paths between the source node and the destination, source node needs to have the right path to select algorithm, and this algorithm needs to consider the specific application requirements, quality of service, transmission reliability and delay, etc. In real networks, data transmission between two nodes usually go through wired and wireless networks, and wired networks path diversity is not within the scope of this study.

This realization process of path diversity transmission and the protocol it used are shown in Figure 6, the source node in the figure has a wired network interface, the IP address is IP, the destination node has three wireless network interface, they are WLAN, GSM and 3G, and their IP addresses are IP_d^1, IP_d^2, IP_d^3. When the encoder generates a video stream, deal the video stream packet using the PFDA algorithm. This algorithm encapsulates and sends the packet by calling the RTP protocol, and specifies the IP address of source node and the destination node in the interface function, one of the end to end path between two nodes in this figure can be expressed as $p_1 : \{IP_s, IP_d^1\}$. The End to End Virtual Path described in the text is formed by one or more paths in the above end to end path, which means all paths compose nonempty subset of a collection, and it can be expressed as:

$$VP \subseteq \{p_1, p_2, p_3\} \cap VP \neq \varphi$$

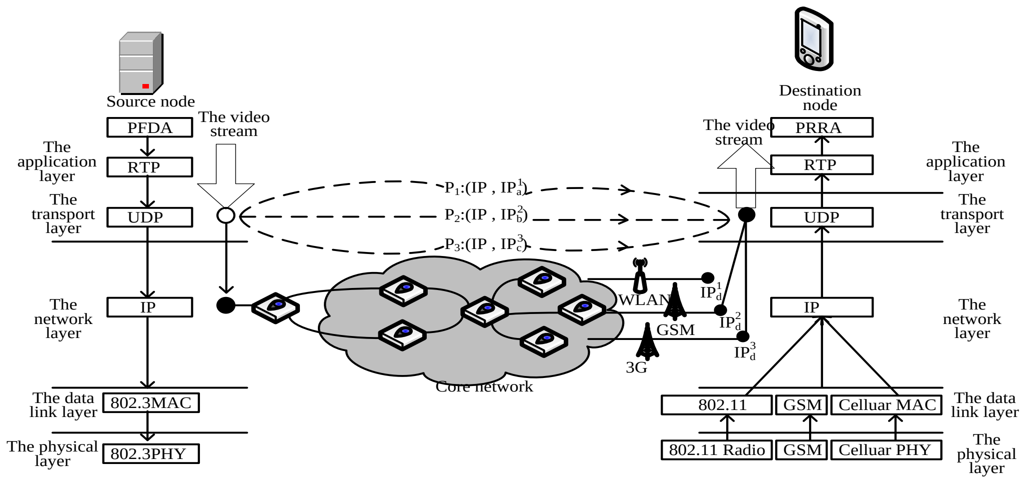


Figure 6. Realization process of path diversity transmission and the protocol

Table 1. Comparison of EDPE algorithm with PFDA-PRRA algorithm

Algorithm name	Type	Achievement	Path parameter	Length
EDPF	Network layer	Network Proxy Server	Delay Broadband	Length weights
PFDA-PRRA	Application laye	Streaming media server client	Delay Broadband	SVC scalability

It can be drawn from the above process that each protocol layer below RTP does not use PFDA-PRRA algorithm, so this paper fulfill path diversity transmission and build end to end virtual channel in the application layer.

3. PFDA-PRRA Algorithm

PFDA-PRRA algorithm is need to effective integration the bandwidth, and at the same time there are many disorder problems caused by multi-path transmission of data packets need to be considered, because of the channel state reasons (such as bandwidth, delay). In the first section, there is a summary on these several paths diversity transmission packet scheduling algorithm, including EDPF, LBA and OPSSA. In the above three algorithms, EDPF is consistent with the purposed algorithm, that is, the delay is minimized; table 1 lists type, implementation, the considered path and the data packet parameters of the proposed algorithm PFDA-PRRA, and then they are compared with EDPF.

3.1. PFDA Algorithm

The purpose of designing the server data packet scheduling algorithm is to minimize the one-way packet delay within the maximum transmission capacity. The algorithm is named as packet fragment transmission algorithm, in which it is specifically divided into the following three steps: selective packet loss, decoding time calculation and packet decode segment transmission.

(1) Selective loss. Server according to a certain time interval $t_{interval}$ sends packets. Firstly, the total length of video streams (maximum transmission capacity) is determined in the interval. If the length is less than the total

length of the packet, selective packet discard is needed to choose. This can save unnecessary bandwidth overhead, but also can decline the quality of the video to minimum. The total delay of individual data packets through end to end in the path is equal to transmission delay combined with the propagation delay. Calculation algorithm is as follows:

$$e^{total} = e^{trans} + e^{prop} = t_{pkt} / s\varpi + e^{prop} \quad (1)$$

Wherein, t_{pkt} indicates that the length of data packet; $s\varpi$ and e^{prop} are respectively represent the bandwidth and propagation delay of the path. Therefore, the maximum transmission capacity $s_{\text{int}\,erval}$ of the path i within the $WR^{i}_{\max\_trans}$ is calculated as follows

$$WR^{i}_{\max\_trans}(s_{\text{int}\,erval} - e_i^{prop}).s\varpi_i \quad (2)$$

Then all the paths in the terminal equipment can be drawn and conduce the maximum transmission capacity:

$$WR_{\max\_trans} = \sum_{i=1}^{E_{net}} WR^{i}_{\max\_trans} = \sum_{i=1}^{E_{net}} (s_{\text{int}\,erval} - e_i^{prop}).s\varpi_i \quad (3)$$

After that, the selective loss is needed to conduct according to the weight and length in the $A(p)$ data packet. Packet weight p^{ϖ} need to read packets to determine because the decoding of enhancement layer data packets need to refer to the base layer, so that its weight is less than the base layer data packets. Discarded first enhancement layer packets do not affect the underlying layer decoding, basic video quality can be ensured. So the packets are dropped one by one according to the general p^{ϖ} values from small to large, until the sum of A (P)'s remaining data packet length is no greater than $WR_{\max\_trans}$, namely:

$$WR_{\max\_trans} \ge \sum_{i=1}^{N_{A(p)}} P_i^{size}$$

(2) Decoding time calculations. After the selective loss is over, the remaining data packet in $A(p)$ is the required time within intervals of the transmitted packet. Because there are constraints of play back delay caused by the client Δ and the maximum delay $WR_{\max}$, each data packet's decoding time needs to be individually calculated and the normal decoding time of the packet is equal to the play time of the data packet, and it is calculated as follows:

$$p^{d} = p^{seq} / \delta + \Delta \quad (4)$$

Wherein, δ represents the play speed, but usually the maximum delay limit is less than play time, hence the decoding time required to take the minimum between the two:

$$p^{d} = MIN(p^{seq} / \delta + \Delta, DB_{\max}) \quad (5)$$

(3) Packet fragments sent. According to RTP specification of $H.264$, the packet sending need to follow the coding sequence, namely the coding sequence; After the decoding time calculations are complete, the packet in $A(p)$ in accordance with the coding sequence number of packets gets the order and the fragmented packets are individually identified. Because $H.264RTP$ specification does not allow that some packets are be fragmented, such as parameter sets and Supplemental Enhancement Information packet. Such kind of data packet length is small, and there is no need to consider the segments. Their approach is to directly select an end to end delay shortest path to send the message. Path selection uses the Pekingese q, and the expression is shown as follows

$$p^{q} = i, 1 \le i \le N_{net} \quad (6)$$

Segments steps need to calculate the number of sub-segments, the length of each segment and the transmitted path. To take advantage of bandwidth in the wireless network connected by the mobile terminal, the number of segments is assumed as the number N in wireless network, the packet's segmentation vector expression is as follows:

$$\begin{gathered} p = (p_1, p_2, ..., p_i, ..., p_{N_{net}}) \\ Subject\ to \sum_{i=1}^{N_{net}} p_i^{size} = p^{size} \end{gathered} \quad (7)$$

Because all segments path is diversity transmission via the client only after the normal reorganization decode the packet is segmented end delay is equal to the maximum value of all the sub-end delay, each path segment length transmission purposes is calculated to minimize delay, the expression is as follows

$$\begin{aligned} &Minimize: s^{m} = MAX\left\{s_i^{size} / s\varpi_i + d_i^{prop}\right\}, 1 \le i \le N_{net} \\ &Subject\ to \sum_{i=1}^{N_{net}} s_i^{size} = s^{size} \end{aligned} \quad (8)$$

The converted expression for the equation is shown as following

$$\begin{cases} s_i^{size} / s\varpi + s_i^{prop} = s_{i+1}^{size} / s\varpi_{i+1} + s_{i+1}^{prop}, 1 \le i \le N_{net} - 1 \\ \sum_{i=1}^{N_{net}} s_i^{size} = s^{size} \end{cases} \quad (9)$$

The results can be obtained from solving equations

$$s_i^{size} = \left[s^{size} - \sum_{k=1}^{N_{net}} s\varpi_k . \left(s_i^{prop} - s_j^{prop}\right)\right] . \left(s\varpi_i / \sum_{i=1}^{N_{net}} s\varpi_i\right) \quad (10)$$

s_i^{size} is the segment length of the transmission path; pseudo-code description of the PFDA algorithm is shown as in Algorithm 1.

Algorithm1. Packet Fragment transmit the algorithm PFDA

$Initialize: k \leftarrow p; Acculen \leftarrow k;$

$$WR_{\max\_trans} \leftarrow \sum_{i=1}^{N_{net}} (t_{initral} - d_i^{prop}).s\varpi_i;$$

1) $IF\ WR_{\max\_trans} \prec \sum_{i=1}^{N_{net}} p_i^{size}\ then$

2) $SORT = BY - WEIGHT(A(P))$

3) $p = (p_1, p_2, ...p_3, ...pN_{net})$

4) $WHILE\ j \prec N_{net}\ and\ DiscardSize \succ j\ DO$

5) $DiscardSize \leftarrow \sum_{i=4}^{N_{net}} p_i^{size} - WR_{\max\_trans}$

6) $N_{net} \leftarrow N_{net} - 1$

7) $j \leftarrow j + 1$

8) $ENDWHILE$

9) $END\ IF$

10) $SORT - BY - SEQNUM(A(P))$

11) $FOR\ EACH\ packet\ p\ in\ A(p))$

12) $p^4 = \min(p^{size} / \sigma + \vartriangle, sw_{\min})$

13) $IF\ IS - PACKET - RECEIVED(p^{prop}) == ture\ THEN$

14) $FOR\ i \leftarrow 1\ TO\ N_{net}$

15) $IF\ i = N_{net}\ then$

16) $p_i^{size} \leftarrow \left[p^{prop} - \sum_{k=1}^{N_{net}} s\varpi_i.(d_i^{pop} - d_j^{pop}) \right].(s\varpi_i / \sum_{i=1}^{N_{net}} s\omega_i)$

17) $p_i \leftarrow PACKET - FRAGMENTE(p.AccuLen, P_i^{size})$

18) $AccuLen \leftarrow AccuLen + p_i^{size}$

19) $ELSE$

20) $p_1 \leftarrow PACKET - FRAGMENTE(p.AccuLen)P^{SIZE-}AccuLen$

21) $P_i^e \leftarrow i$

22) $END\ IF$

23) $ENF\ FOR$

24) $ELSE\ P^W \leftarrow \underset{1 \le I \le N_{net}}{MIN}(p^{size} / s\varpi_i + d_i^{size})$

25) $END\ FOR$

3.2. PRRA Algorithm

Packet after the segmentation and transmission via the server needs to be restructured in the client decodes the transmission process, there will often packet loss, out of order, the delay is longer, etc., in order to solve these problems will decode rate maximization, this paper designs a client packet reception and restructuring algorithm, named PRRA, the algorithm can be divided: Packet validation, caching and cache receives adaptive sliding window. Data structure using packet to receive multi value mapping, namely a primary key (key) correspond to multiple values (value), and its coded serial number as the primary key, in which the primary key is corresponding to three values, namely the total packet length, received length and packet pay load shown in Figure 7

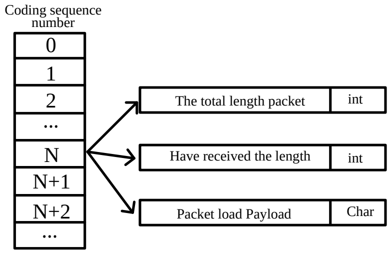


Figure 7. Structure diagram of packet receipting the cache data

When a packet is received after Pekingese, in the RTP header the decoding time p^d can be obtained, while the coding sequence number, the total length of belonged packet (NAL unit), the starting position of the segment and length of the section can be got from the header from the custom, and the remaining information like weight p^{ϖ} of a packet need to be read from packets load. After the above information is read, the specific steps into the The pseudo-code of PRRA algorithm is shows as below.

Algorithm 2 Reorganization algorithm PRRA of packet reception

$Initialize: LoopFlag \leftarrow true;$

1) $IF\ s^{cum} - p^d\ or\ p^{seq} \le DecSeq\ then$

2) $DISCARE - PACKET(t)$

3) $ELSE\ FOR\ EACH\ Packet\ p'\ in\ D(t)$

4) $IF\ IS - DISCARDED(T')\ then$

5) $DISCARD - PACKET(t)$

6) $END\ IF$

7) $END\ FOR$

8) $END\ IF$

9) $IF\ IS - PACKET - RECEIVED(t^{seq}) == ture\ then$

10) $t_b \leftarrow RETURN - BY - SEQNUM(B(t), t^{seq})$

11) $t_b \leftarrow COMBINE - PACKETS(t_b, t)$

12) $t_b^{size} \leftarrow t_v^{size} + t^{size}$

13) $ELSE\ ADD - PACKET(B(p), p)$

14) $END\ IF$

15) $WHILE\ LoopFlag\ DO$

16) $IF\ t_{temp} == NIL\ THEN$

17) $IF\ t^{tur} \prec DecSeq / \varsigma + \vartriangle then$

18) $LoopFlag \leftarrow false$

19) $ELSE\ DecSeq \leftarrow false$

20) $ELSE\ DecSeq \leftarrow DecSeq + 1$
21) $END\ IF$
22) $ELSE\ IF\ T^{CUR} - P_{temp}^{cut} \prec p_{temp}^{d}\ then$
23) $PROCESS - ACCESSUNIT(t_{temp})$
24) $DecSeq \leftarrow DecSeq + 1$
25) $ELSE\ LoopFlag \leftarrow false$
26) $END\ IF$
27) $ELSE\ REMOVE - PACKET(B(p)), p_{temp})$
28) $DecSeq \leftarrow DecSeq + 1$
29) $END\ IF$
30) $END\ IF$
31) $END WHILE$

4. Two Experimental Simulation and Analysis

4.1. Experimental Environment

(1) The video codec uses the reference software Joint Scalable Video Model (JSVM9.12) 1)Of H. 264/SVC, and the bit rate respectively use 1Mbps and 3Mbps.

Standard video sequence $Forema\_cif$ with 300 frames is used as the encoding YUV file. The main encoding parameters and its sets are shown in Table 2.

Table 2. Main parameters

Parameter	Set
Image length	4 frames
Frame rate	32fps
Stratified number	2
Quantization parameter(Layer 0)	30
Quantization parameter(Layer 1)	36

(2) Network simulation tool uses QualNet 2), which is a powerful mainstream network emulator. But application layer on QualNet protocol does not have video transmission capabilities. So this article makes JSVM code in the form of a static link library, combines with QualNet and adds video codec and transport protocol. Main parameters are shown in Table 3. Experimental scenario is shown in Figure 8.

Table 3. Main parameter setting

Parameter	Set up
Bandwidth(Path 1)	350kbps
Bandwidth(Path 2)	200kbps
Bandwidth(Path 3)	150kbps
Node mobility model	RandomWaypoint Model
interval of transmission time	1.5s

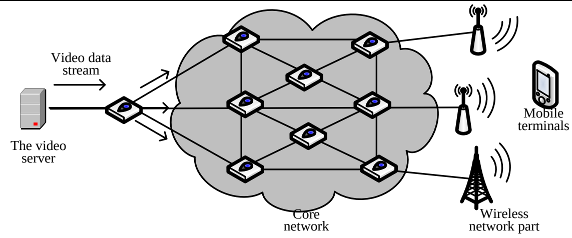


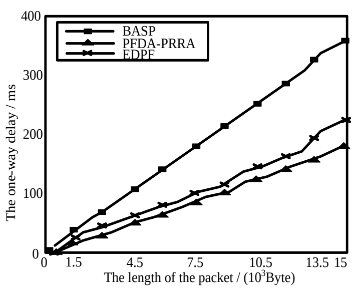
Figure 8. Experiment scene topology

4.2. Results Analysis

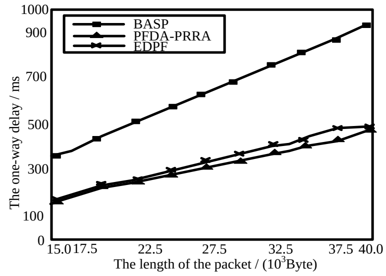
This paper uses three comparison algorithms: PFDA, PRRA, EDPF and Bandwidth Aggregated Single Path (BASP). Bandwidth integrated single path represents the ideal state of integrated heterogeneous wireless network bandwidth. Therefore, compared with the above three algorithms, the proposed integrated bandwidth of PFDA and PRRA algorithms can be examined and the performance of one-way delay can be reduced. The used

One-way delay In the section 2.1, it has been given the length distribution range and probability of two different bitrates packet. In order to ensure the adequate sampling, when the bit rate is equal to 1Mbps, the sampling interval is as 0 ~ 15KB; while when the bit rate is equal to 3Mbps, the sampling time range is as 15 ~ 40KB. The results are shown in Figure 9.

Figure 9 (a) can be seen, when the packet length is small, the delay of PFDA and PRRA is slightly larger than the delay of BASP; it is because the data packets are fragmented, and the factors of the custom header length attached and network transmission random. In the measuring samples of the experiments, the delay of these two algorithms are very close; the trend is very clear in Figure 9 (b), in which the individual large differences are existed(e.g. when the packet length is equal to 32515 bytes , the delay of BASP is 355.6ms, while the delay of PFDA and PRRA is 388.56ms). Therefore, it can be concluded: the advantages of PFDA and PRRA way in reducing the delay are obvious compared with the EDPF, which is very close to the ideal bandwidth aggregation state BASP.



（a）BitRate=1 Mbps



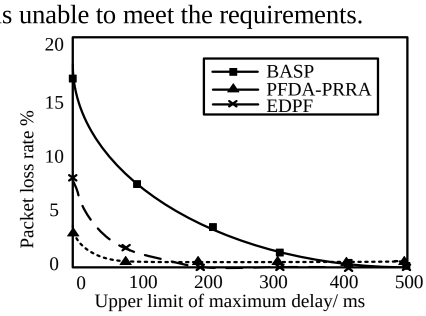
（b）BitRate=3 Mbps

Figure 9. One-way delay

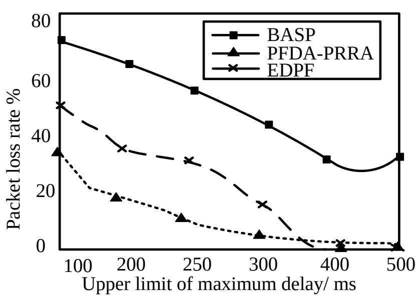
Packet loss rate The results are shown in Figure 10; when the upper limit is given, the packet loss rates of the PFDA and PRRA and BASP are very close with two particular bit rates, which is significantly lower than EDPF. From the experimental results, it can draw the following two conclusions:

1) When the bit rate is equal to 1Mbps, and when the maximum limit is equal to 150ms, the delay loss rates of BASP and PFDA and PRRA respectively are 5.35% and 5.69%. Basically this can achieve a good quality of service requirements; at this moment the packet loss rate of EDPF is 25.1%, which can not provide a good quality of service. By used EDPF, when maximum delay limit as maximum of 400ms, it can reach the requirement of acceptable quality of service, and then packet loss rate of EDPF is 1%.

2) When the bit rate is equal to 3Mbps time, PFDA and PRRA can satisfy the acceptable quality of service; when the maximum delay time is 400ms, the packet loss rate is 7.34%, while the packet loss rate of EDPF is 29.8%, which is unable to meet the requirements.



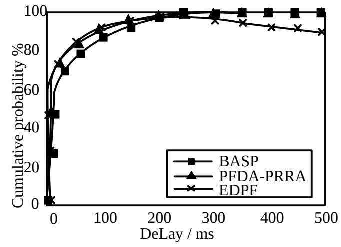
（a）BitRate=1 Mbps



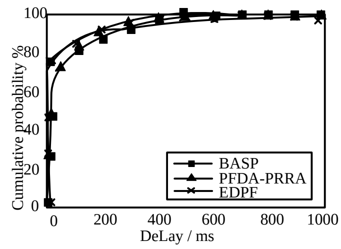
（b）BitRate=3 Mbps

Figure 10. Loss rate

Delay cumulative distribution. The results are shown in Figure 11; when the ratio is equal to 1Mbps, BASP, PFDA and PRRA respectively are 98.1% and 94. 92% of the packet one-way delay is within 150ms, but this time EDPF is only as 91. 57%. When the bit rate is equal to 3Mb, the one-way delay of the BASP and PFDA and PRRAS are 99.5% and 96.5% t within the 400ms, while EDPF is only occupied 89.9%.



（a）BitRate=1 Mbps



（a）BitRate=3 Mbps

Figure 11. Delay Cumulative distribution function

5. Conclusion

This paper presents an end-to-end service quality assurance algorithm of mobile multimedia conferencing in heterogeneous wireless network. For transmission one-way delay of multimedia video, the algorithm combines the packets fragmenting and restructuring with the transmission path diversity in heterogeneous wireless network; in considering the case of bandwidth, delay, length, weight and other parameters, the packets are fragmented and send. Each segment is transmitted through path diversity to reaches the client, and the effective validation and restructuring are conducted. Simulation results show that the algorithm can effectively integrate the bandwidth of heterogeneous wireless network and reduce the one-way delay of video streaming. In the future the further study will be conducted aimed at channel errors, packet loss, bandwidth fluctuations, packet disorder and other issues.

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